L201 Voip Gateway

User's Manual

– Version: 2.33 –

Table of Contents

TABLE OF CONTENTS 2

PREFACES 5

0.1 ABOUT THIS MANUAL **5**
0.2 COPYRIGHT DECLARATIONS **5**
0.3 TRADEMARKS **5**
0.4 SAFETY INSTRUCTIONS **5**
0.5 WARRANTY **5**

INTRODUCE 6

1.1 OVERVIEW **6**
1.2 ACRONYMS TABLE **6**
1.3 INTRODUCTION **7**
1.4 FRONT PANEL LED INDICATORS & REAR PANELS **8**
1.4.1 FRONT PANEL LED AND CONTAINER DESCRIPTIONS **8**
1.4.2 REAR PANEL DESCRIPTIONS **8**
1.5 FEATURES AND SPECIFICATIONS **10**
1.5.1 VOIP GATEWAY FEATURES **10**

INSTALLATION AND SETUP 10

2.1 PACKAGE CONTENT **10**

2.2 INSTALLATION **11**
2.3 SETUP **14**
2.3.1 FACTORY DEFAULT SETTING **14**
2.3.2 TELNET **14**
2.3.3 WEB USER INTERFACE **17**

WIZARD FOR QUICK SETUP 21

3.1 WAN PORT TYPE SETUP **21**

3.2 CONFIGURING NAT OR BRIDGE SETTING: **23**

3.3 VOIP CALL PROTOCOL SETUP **24**

GATEWAY SETTING 25

4.1 NETWORK CONFIGURATION **27**
4.1.1 WAN PORT TYPE SETUP **27**
4.1.2 CONFIGURING LAN IP ADDRESS AND DHCP SERVER **29**
4.1.3 VIRTUAL SERVER SETUP **30**
4.1.4 DYNAMIC DNS **30**
4.1.5 NETWORK MANAGEMENT **31**

4.2 VOIP SETUP **32**
4.2.1 H.323 SETUP **32**
4.2.2 SIP SETUP **40**
4.2.3 DIRECT CALL (PEER TO PEER) SETUP **46**
4.2.4 OTHER VOIP SETTING **49**

4.3 SYSTEM ADMINISTRATOR **51**
4.3.1 SAVE CONFIGURATION AND REBOOT **51**
4.3.2 ACCESS CONTROL **51**
4.3.3 SET TO DEFAULT CONFIGURATION **52**
4.3.4 BACKUP/RESTORE CONFIGURATION TO A FILE **52**
4.3.5 SYSTEM INFORMATION DISPLAY FUNCTION **53**
4.3.6 SNTP SETTING FUNCTION **53**
4.3.7 Syslog FUNCTION **54**
4.3.8 Capture Packets Function **54**

4.5 UPDATE FIRMWARE **55**

APPENDIX 58

A FAQ LIST **58**

B SIP SETTING VOIPBUSTER **60**

C ANSWER SUPERVISION **62**

D SIP SPEEDS CALL **64**

E INTEROPERABILITY LIST **66**

F GATEWAY VALUE SETTING **67**

G SCENARIO APPLICATION SAMPLES **69**

PREFACES

0.1 About This Manual

This manual is designed to assist users in using the VoIP Gateway. Information in this document has been carefully checked for accuracy; however, no guarantee is given as to the correctness of the contents. The information contained in this document is subject to change without notice.

0.2 Copyright Declarations

Copyright 2006 Telephony Corporation. All rights reserved. This publication contains information that is protected by copyright. No part may be reproduced, transmitted, transcribed, stored in a retrieval system, or translated into any language without written permission from the copyright holders.

0.3 Trademarks

Products and Corporate names appearing in this manual may or not be registered trade marks or copyrights of their respective companies, and are used only for identification or explanation and to the owners' benefit, without to infringe.

0.4 Safety Instructions

The most careful attention has been devoted to quality standards in the manufacture of the Gateway. Safety is a major factor in the design of every set. But, safety is your responsibility too.

- **Use only the required power voltage. Power Input: AC 100-240V, 50-60Hz**
- **To reduce the risk of electric shock, do not disassemble this product. Opening or removing covers may expose the Gateway to hazardous voltages. Incorrect reassembly can cause electric shock when this product is subsequently used.**
- **Never push objects of any kind into the equipment through housing slots since they may touch hazardous voltage points or short out parts those could result in a risk of electric shock. Never spill liquid of any kind on the product. If liquid is spilled, please refer to the proper service personnel.**
- **Use only Unshielded Twisted Pair (UTP) Category 5 Ethernet cable to RJ-45 port of the Gateway.**

0.5 Warranty

We warrant to the original end user (purchaser) that the gateway will be free from any defects in workmanship or materials for a period of one (1) years from the date of purchase from the dealer. Please keep your purchase receipt in a safe place as it serves as proof of date of purchase. During the warranty period, and upon proof of purchase, should the product have indications of failure due to faulty workmanship and/or materials, we will, at our discretion, repair or replace the defective products or components, without charge for either parts or labor, to whatever extent we deem necessary to re-store the product to proper operating condition. Any replacement will consist of a new or re-manufactured functionally equivalent product of equal value, and will be offered solely at our discretion. This warranty will not apply if the product is modified, misused, tampered with, damaged by an act of God, or subjected to abnormal working conditions. The warranty does not cover the bundled or licensed software of other vendors. Defects which do not significantly affect the usability of the product will not be covered by the

warranty. We reserve the right to revise the manual and online documentation and to make changes from time to time in the contents hereof without obligation to notify any person of such revision or changes.

Note

Repair or replacement, as provided under this warranty, is the exclusive remedy of the purchaser. This warranty is in lieu of all other warranties, express or implied, including any implied warranty of merchantability or fitness for a particular use or purpose. We shall in no event be held liable for indirect or consequential damages of any kind of character to the purchaser.

To obtain the services of this warranty, contact us for your Return Material Authorization number (RMA). Products must be returned Postage Prepaid. It is recommended that the unit be insured when shipped. Any returned products without proof of purchase or those with an out-dated warranty will be repaired or replaced and the customer will be billed for parts and labor. All repaired or replaced products will be shipped by us to the corresponding return address, Postage Paid. This warranty gives you specific legal rights, and you may also have other rights that vary from country to country.

Introduce

L series VoIP Gateways are the low to high VoIP total Solutions. This document describes the usage of the Voice gateway.

1.1 Overview

- VoIP Gateway which is a device that allows one to connect a normal PSTN telephone to the Internet in order to make or place telephone calls.
- VoIP Gateway device may work in conjunction with a computer, such as an IP-sharing / Router, or it may be a stand-alone device that communicates with a service provider over the Internet.
- VoIP Gateway provides a direct analog interface for computer modems, fax machines, analog telephones, and other devices that require an analog port.
- 2/4/8 port Series VoIP Gateway can build in a simple H.323 Gatekeeper or SIP Proxy Server.
- VoIP Gateway also support standard Internet services, such as IP-Sharing, NAT, Virtual Server, DDNS, QOS, Port Filter, IP Filter function.

1.2 Acronyms Table

Acronym:	Full Name:	Acronym:	Full Name:
ADC	Analog to Digital Converter	CODEC	Coder / Decoder
DAC	Digital to Analog Converter	DC	Direct Current
DDNS	Dynamic Domain Name System	DHCP	Dynamic Host Configuration Protocol
DMZ	Demilitarized Zone	DNS	Domain Name System
DTMF	Dual Tone Multi Frequency	FXO	Foreign Exchange Office
FXS	Foreign Exchange Station	GMT	Greenwich Mean Time
IP	Internet Protocol	IPsec	Internet Protocol Security
L2TP	The Layer 2 Tunnel Protocol	LAN	Local Area Network
WAN	Wide Area Network	MAC	Media Access Control
MII	Media Independent Interface	NAT	Network Address Translation
NTP	Network Time Protocol	PPTP	Point-to-Point Tunneling Protocol
RTP	Real-Time Transport Protocol	RTCP	Real-Time Transport Control Protocol (also known as RTP

			control protocol)
SIP	Session Initiation Protocol	SLIC	Subscriber Line Interface Circuit
STUN	Simple Traversal of UDP through NATs	URI	Uniform Resource Identifier
TCP	Transmission Control Protocol	UDP	User Datagram Protocol
UPnP	Universal Plug and Play	VoIP	Voice Over Internet Protocol

1.3 Introduction

This VoIP Gateway provides a total solution for integrating voice-data network and PSTN.

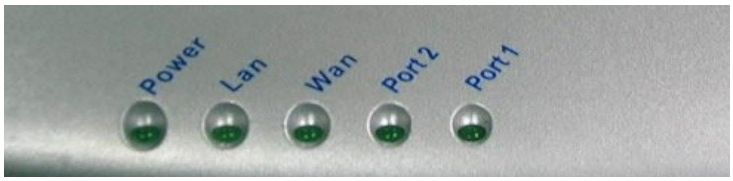
The L200 gateway is a low to high density port gateway which support SIP / H.323 VoIP Protocol. Low model (2/4 port) can embedded H.323 Gatekeeper or SIP Proxy Server (Option). **The L200** gateway allows 2 lines analog voice and fax communication over a traditional data communications/data networking digital Internet.

1.4 Front Panel LED Indicators & Rear Panels

L200 Series:
-L200 Series Gateway

1.4.1 Front Panel LED and Container Descriptions

L200(GW/GK/SVR) Series

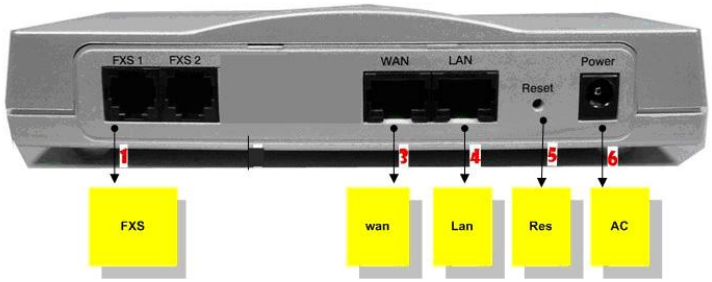


LED	State	Description
1. POWER	On	ATA is power ON
	Off	ATA is power Off
2. WAN port	On	ATA network connection established
	Flashing	Data traffic on cable network
	Off	Waiting for network connection
3. LAN port	On	LAN is connected successfully
	Flashing	Data is transmitting
	Off	Ethernet not connected to PC
4. FXS	Off	Telephone Set is On-Hook
	Flashing	Ring Indication
	On	Telephone Set is Off-Hook
5. FXO	Off	Line is On-hook
	On	Line is In-Use

NOTE: System initialization will turn some LEDs ON for a few sec.
When System Boot/Reboot , the Port LEDs will flash in turn for a few sec.

1.4.2 Rear Panel Descriptions

L200 Series



Item	Port	Description
1	FXS(Foreign Exchange Station)	FXS port can be connected to analog telephone sets or Trunk Line of PBX.
2	FXO(Foreign Exchange Office)	Can be Connected to PBX or CO line with RJ-11 analog line. FXO port can be connected to the extension port of a PBX or directly connected to a PSTN line of carrier.
3	WAN(Wide Area Network)	Connect to the network with an Ethernet cable. This port allows your ATA to be connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
4	LAN(Local Area Network)	Connect to PC with Ethernet cable. 1 port allows your PC or Switch/Hub to be connected to the ATA through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
5	RES(Reset button)	Push this button until 3 seconds, and ATA will be set to factory default configuration.
6	AC power(DC in 12V)	A power supply cable is inserted

1.5 Features and Specifications

The **L200** Series Gateways provide many built-in server and software features to provide a convenient comprehensive solution for your VoIP network

1.5.1 Gateway Features

VoIP Key Features

- Both support SIP and H.323 protocols: SIP Registration and Digest Authentication and H.323 Gatekeeper Registration.
- Single Number / Account for multiple ports.
- Smart VoIP call Dialing Book: VoIP call Book could provide any application VoIP call to any type destination (Domain name / IP address, PSTN or PBX) or hunting number setting.
- AC termination Impedance : 600/900 OHM and complex impedance
- Answer Supervision for Polarity Reversal Detection and Voice detection
- NAT traversal: This feature allow gateway to operate behind any NAT/Firewall device. There is no need to change any configuration of NAT/Firewall like setting virtual server.
- Smart-QoS: This feature provides good voice quality when user place a VoIP call and access internet at the same time. The gateway will automatically start to reserve bandwidth for voice traffic when VoIP call proceeds.
- Call Hunting Facility: This function helps gateway to use the lines effectively. This facility automatically transfers your incoming call to a free line. Subscribers need not indicate numerous numbers of each port of gateway.
- Voice channels status display: This function display each port status like as on-hook, off-hook, calling number callee's number, talk duration, codec.
- Flash Detection and Generation Program: FXO support Flash Generation and FXS support Flash Detection.
- CDR: Use Syslog Server to receive CDR information that gateway send by UDP.

Call Features

- Voice channels status display
- Direct Dialing Mode : peer to peer call (support IP Address Call or Domain Name Call)
- Register Call Mode : register to SIP Proxy Server or H.323 Gatekeeper
- Adjustable volume : - 9 db ~ 9 db
- Silence Compression(VAD)
- Auto Dial for speed
- Dynamic Jitter Buffer
- Hot-Line Support(PLAR)

Configuration & Management

- Web-based Graphical User Interface
- Remote management over the IP Network
- FTP firmware upgrade
- Backup and Restore Configuration file
- Syslog support

Installation and Setup

2.1 Package Content

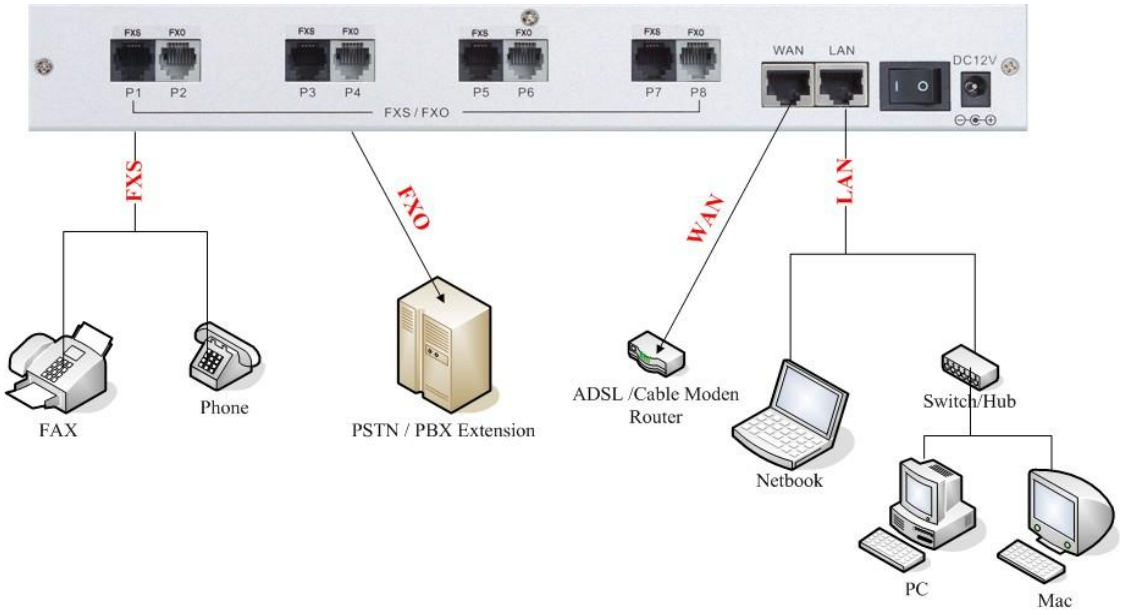
Please check enclosed product and its accessories before installation.

L200 Series Gateway

The 2/4 Port packet contents:	
GW, RJ-45 AC Power Adapter CD-Rom(User manual)	X1 X1 X1 X1

2.2 Installation

Install Gateway
1 Connect the 12V DC IN to the power outlet with power adaptor.
2 Connect FXO to PSTN / PBX Extension Line.
3 Connect FXS to a telephone jack with the RJ-11 analog cable (Phone / PBX Trunk Line.)
Connecting to a PC:
1 Connect the Ethernet cable (with RJ-45 connector) to any LAN port.
2 Connect the other end of the Ethernet cable to your PC's installed network interface card (NIC).
3 The LED indicators at both the Ethernet port and the NIC should be ON.
Connecting to an External Ethernet Hub or Switch:
1 Connect the Ethernet cable (with RJ-45 connector) to WAN port.
2 Connect the other end of the Ethernet cable to DSL/Cable modem or the external Ethernet hub or switch.
3 The LED indicators on both the LAN port and the external Switch.



Port	Description
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WAN(Wide Area Network)	Connect to the network with an Ethernet cable. This port

	allows your GW to be connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
LAN(Local Area Network)	Connect to PC with Ethernet cable. 1 port allows your PC or Switch/Hub to be connected to the GW through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
RES(Reset button)	Push this button until 3 seconds, and GW will be set to factory default configuration.
AC power(DC in 12V)	A power supply cable is inserted

The hardware installation is now complete. The following sections will guide you through setting up your management PC and connecting to the Web User Interface.

2.3 Setup

There are 2 ways to setup the gateway - **[Web User Interface] or [Telnet]**

2.3.1 Factory Default setting

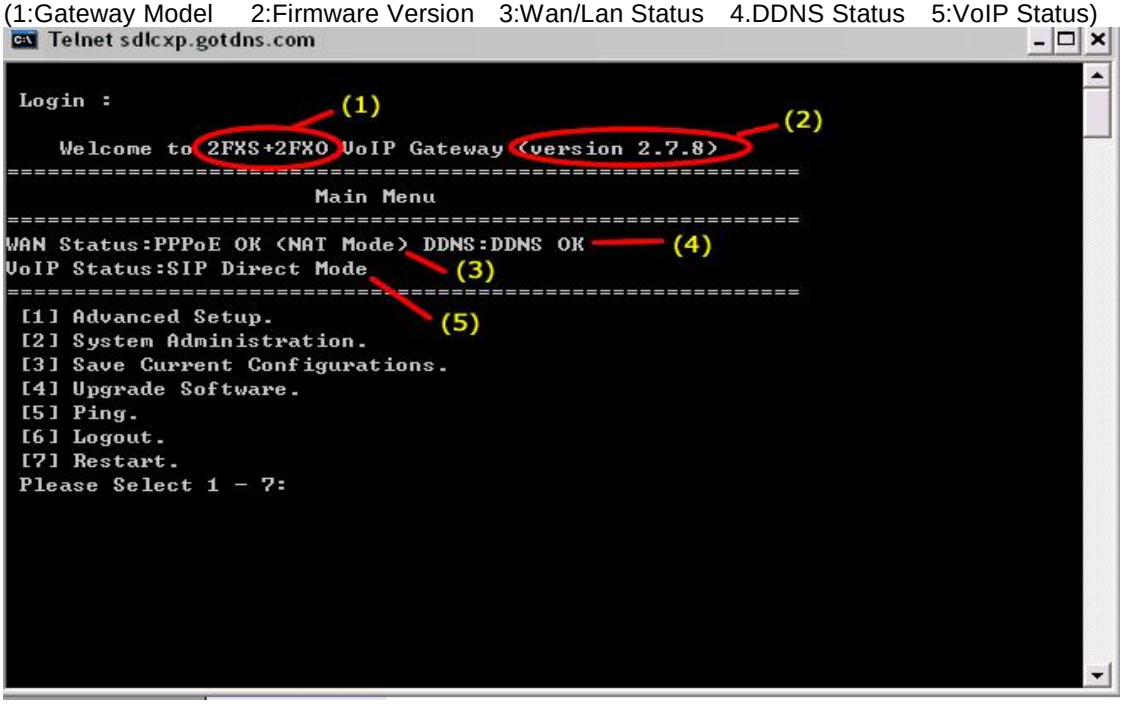
- **WAN** Port IP address : 192.168.1.1
- **LAN** Port IP address : 222.222.222.1
- LAN DHCP Server enable IP range: 222.222.222.51 ~ 222.222.222.100
- **VoIP Number(S200 Series) Port_1~Port_2 number:100,200**
- VoIP default setting was **H.323** signal protocol, **Direct Mode**, **Fast-Start** and **G.723** codec.
- Default login authentication **username : admin, password : admin**

2.3.2 Telnet

Connect WAN/LAN port to Internet or PC and gateway at the same subnet. you can use telnet remote to configure your gateway.

1. Connect Gateway online (Wan/Lan)
2. Remote Gateway by Telnet. If telnet successful, you will see Login display.
(For Example: telnet 222.222.222.1)

3. Input Password (Gateway Access password, Default: admin), If login successful, you will enter the welcome display.
(1:Gateway Model 2:Firmware Version 3:Wan/Lan Status 4.DDNS Status 5:VoIP Status)



4. Gateway Telnet Setting Table, Use 1~9 a~z select setting, “ESC” is back setting.

Item	Setting Option
Main	[1] Advanced Setup. [2] System Administration. [3] Save Current Configurations. [4] Upgrade Software. [5] Ping. [6] Logout. [7] Restart.
[1]Advanced Setup	1.WAN Setting 2.LAN Setting 3.Virtual Server 4.Dynamic DNS 5.Network Management 6.VoIP Basic 7.Dialing Plan 8.VoIP Advance Setting 9.Hot Line Setting a.Port Status
[1]Advanced Setup ……1.WAN Setting	1.Change WAN Type to DHCP 2.Change WAN Type to Fixed IP 3.Change PPPoE Username 4.Change PPPoE Password
[1]Advanced Setup ……2.LAN Setting	1.Change to Bridge Mode 2.Change LAN IP Address 3.Disable DHCP Server 4.Change Start IP Address 5.Change End IP Address 6.Change DNS Server IP 7.Change Lease Time

[1]Advanced Setup ……3.Virtual Server	1.Add Virtual Server 2.Delete Virtual Server
[1]Advanced Setup ……4.Dynamic DNS	1.Change DDNS username 2.Change DDNS password 3.Change DDNS domain name 4.Change DNS server IP
[1]Advanced Setup ……5.Network Management	1.Change web server port 2.Change telnet server port
[1]Advanced Setup ……6.VoIP Basic	1.Change VoIP Protocol to H.323 2.Change Port Number/Account/Password 3.Enable/Disable Public account 4.SIP hunting setting 5.Change SIP Proxy Server IP Address/DNS 6.Use net2phone 7.Change Register Interval(seconds) 8.Enable/Disable SIP authentication 9.NAT Pass Method a.STUN Server address b.SIP realm c.Outbound Proxy Server address d.Change SIP Local Port
[1]Advanced Setup ……7.Dialing Plan	1.Add Outbound Direct Call 2.Delete Outbound Direct Call 3.Add Inbound Direct Call 4.Delete Inbound Direct Call
[1]Advanced Setup ……8.VoIP Advance Setting	**(1)Sip Advance** 1.Set DTMF Relay Mode 2.Change FAX Mode 3.Change RFC2833 Payload(96-127) **(2)Telephone Advance** 1.VAD(Silence Compression)On/Off 2.Change Codec 3.Enable/Disable UK PSTN Tone Detection? 4.Enable/Disable Dial Complete Tone 5.Dial Termination Key Setting 6.FXS Parameters Setting 1.Change FXS Impedance 2.Change Phone In Volume 3.Change Phone Out Volume 4.Flash Detection 5.Ring Frequency 6.FXS Battery reversal generation 7.FXO Parameters Setting 1.Change FXO Impedance 2.Change Line In Volume 3.Change Line Out Volume 4.Change FXO Tx Gain 5.Change FXO Rx Gain 6.Flash Duration 7 DTMF Tone Power 8.FXO Transmit Hybrid 9. Answer Supervision Setting a. Change Ringer Voltage Threshold b. Enable Line Silence Disconnect c. Change FXO Answer delay time d. FXO Ringer Voltage Filter Setting

	(3)Network Advance 1.Disable Smart QOS 2.Bandwidth Control 3.G.723 Bandwidth 4.G.729 Bandwidth 5.Set IP TOS
[1]Advanced Setup ……9.Hot Line Setting	1.Change Port1 Hot Line Number 2.Change Port2 Hot Line Number…….(To your own port)
[2] System Administration.	1.Save Configuration 2.Access Control 3.Set to Default 4.System Information 5.NTP Setting 6.Syslog Setting

2.3.3 Web User Interface

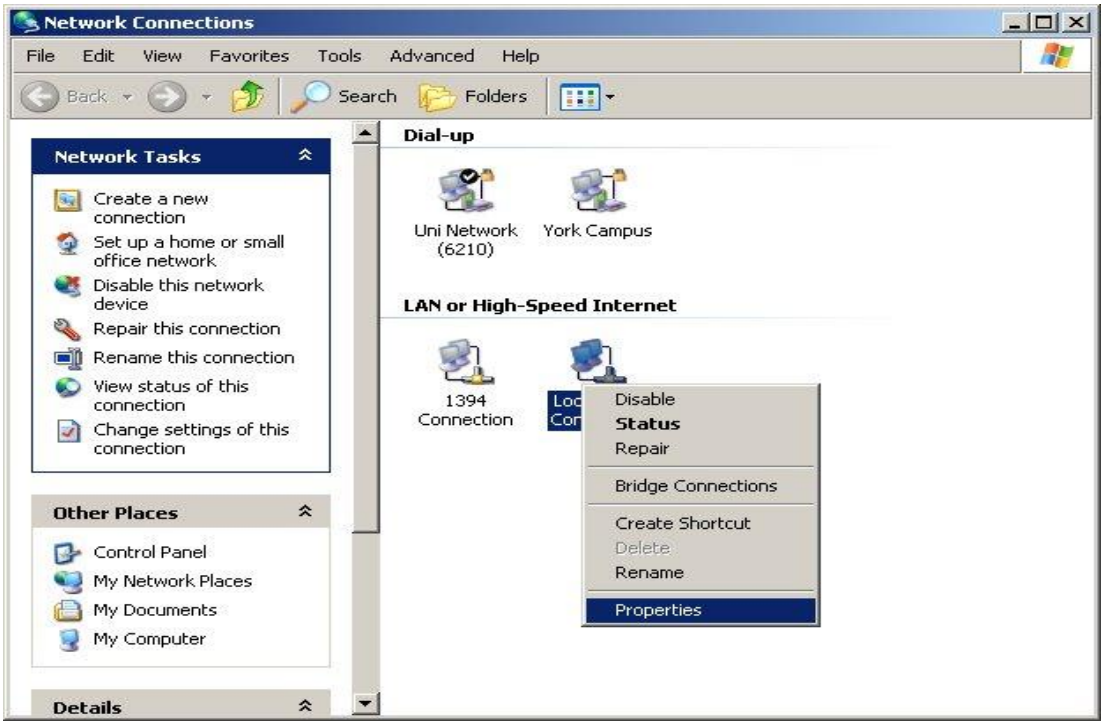
The gateway has a built-in HTTP(Web) server for configuration. Before you use the gateway to access the Internet, you should set up a management PC to log into the router for further configuration. The management PC may be configured with a fixed or dynamically assigned IP address. For a fixed IP address, use an IP address from a 192.168.1.0/24 network, such as 192.168.1.10.

For a dynamic IP address, you need to set the PC as a DHCP client, and then restart or renew the network settings. The DHCP server of router is enabled by default so the PC will then be assigned an IP address and related settings by the router. The following examples are for a MicrosoftTM Windows 2000/XP machine set to use a dynamic IP address.

Checking the Network IP Configuration

The following explains how to setup the Transmission Control Protocol/Internet Protocol (TCP/IP) in Windows 2000/XP. For more detailed information on TCP/IP setup, refer to the Windows 2000/XP help files. For other operating systems refer to the user manuals.

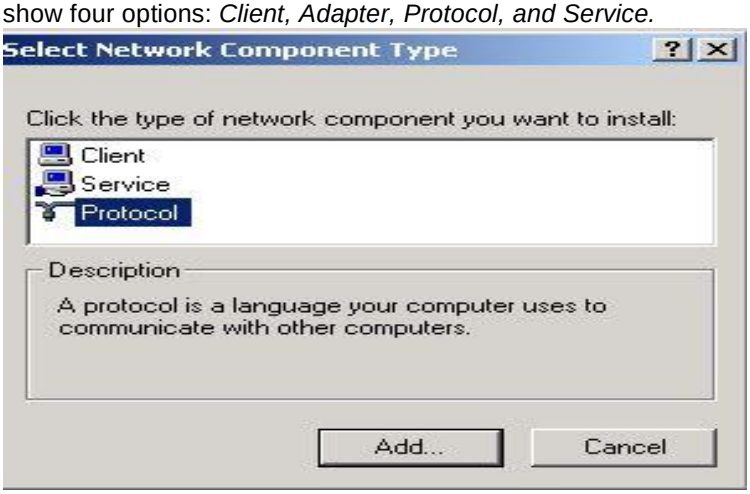
1. On the desktop, Please enter start -> control panel -> network setting.” Click Properties. The Network screen will open.



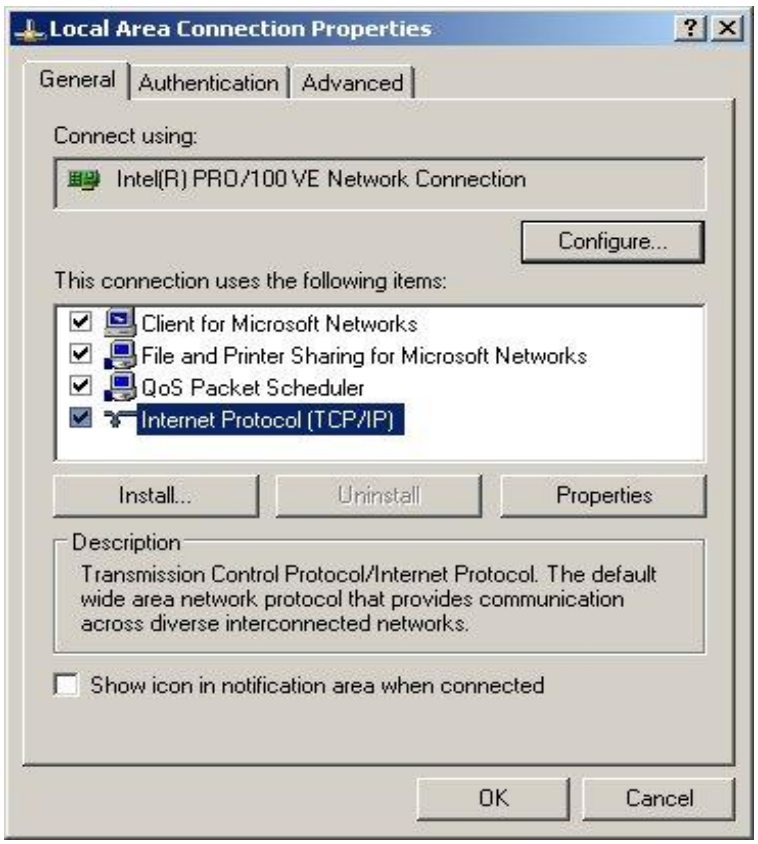
(Your particular system will be different from the screen shown here.)

Check that you have an Ethernet network card installed. If not, refer to the card manufacturer's documentation and install the card and drivers.

If your card is installed,
1. Click the Add button. The Select Network Component Type dialog box will open. The box will show four options: *Client, Adapter, Protocol, and Service.*



2. Select Protocol and click the Add button. The Select Network Protocol dialog box will open.
3. Select Microsoft in the left scrolling window then selects TCP/IP in the right, and click OK.". You will be returned to the Network dialog box.



Configuring the TCP/IP Protocol
1. On the Network dialog box Configuration card, select TCP/IP and then click Properties." The TCP/IP Properties dialog box will open.

2. On the IP Address tab, click Obtain an IP address automatically. As the DHCP (Dynamic Host Configuration Protocol) server built into the router is enabled by default, your computer will get an IP address, subnet mask, and other related IP network settings from the router.

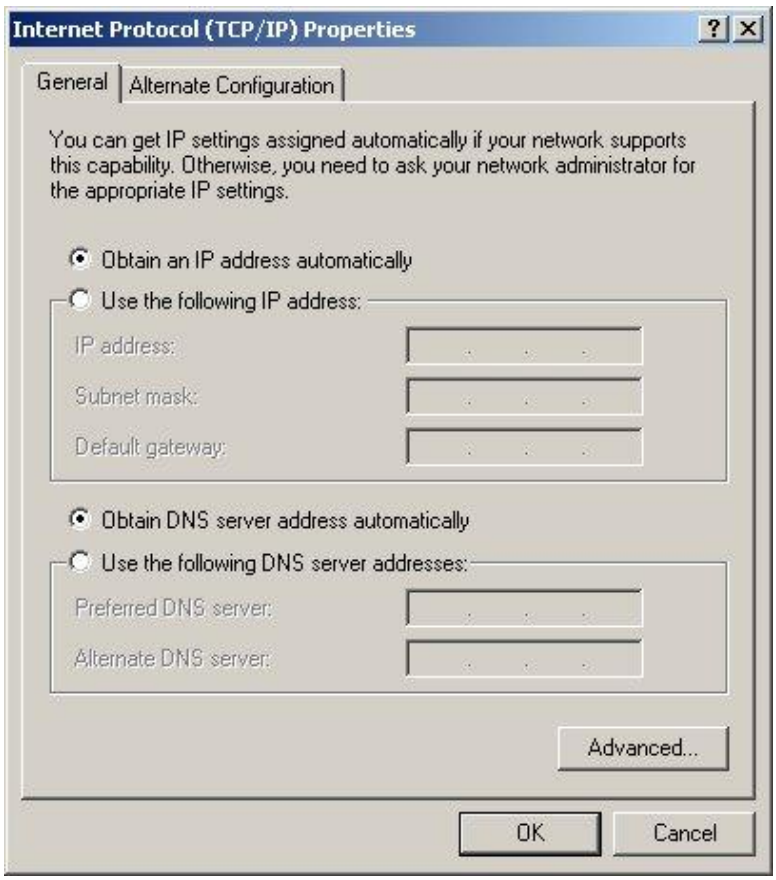
3. On the DNS Configuration tab, click Disable DNS".

4. Click the Gateway tab.

5. Make the New gateway and Installed gateways fields blank and click OK. A dialog box will pop up asking you to restart the PC. Click Yes".

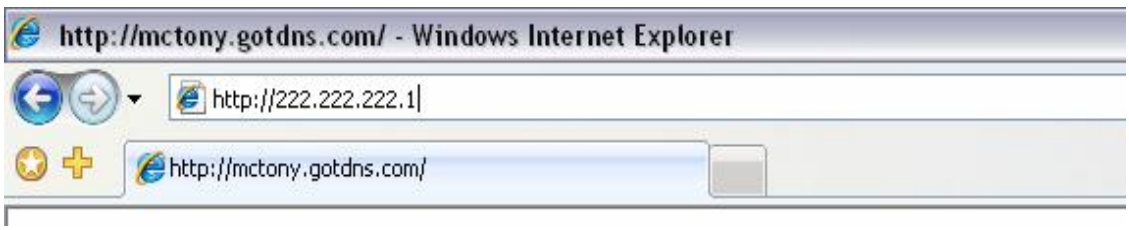
Checking TCP/IP settings
1. After completing the previous steps, click Start -> Run -> and type ipconfig /all. The IP Configuration window will open. If the PC does not show an IP address in the 222.222.222.51 to 222.222.222.100 range, click the ipconfig /release button to release the current configuration. Wait a few seconds and click "ipconfig/renew" to get a new IP configuration from the router.

2. If the IP configuration is correct, you will be able to use the PING diagnostic utility built into Microsoft Windows to ping the router. Click Start -> Programs -> MS-DOS Prompt. A command mode window will open.Type "ping 222.222.222.1" (default IP of the router) to check the network connectivity. If both hardware and software are correct, your computer will receive a response from the router as shown on the next page. If not, verify that the Ethernet cable is connected to the router properly and the Ethernet port LED on the front panel is lit.


Internet Protocol (TCP/IP) Properties
General
Alternate Configuration
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.
Obtain an IP address automatically
Use the following IP address:
IP address:
Subnet mask:
Default gateway:
Obtain DNS server address automatically
Use the following DNS server addresses:
Preferred DNS server:
Alternate DNS server:
Advanced...
OK
Cancel

Connecting to the Web Configuration via a Web Browser

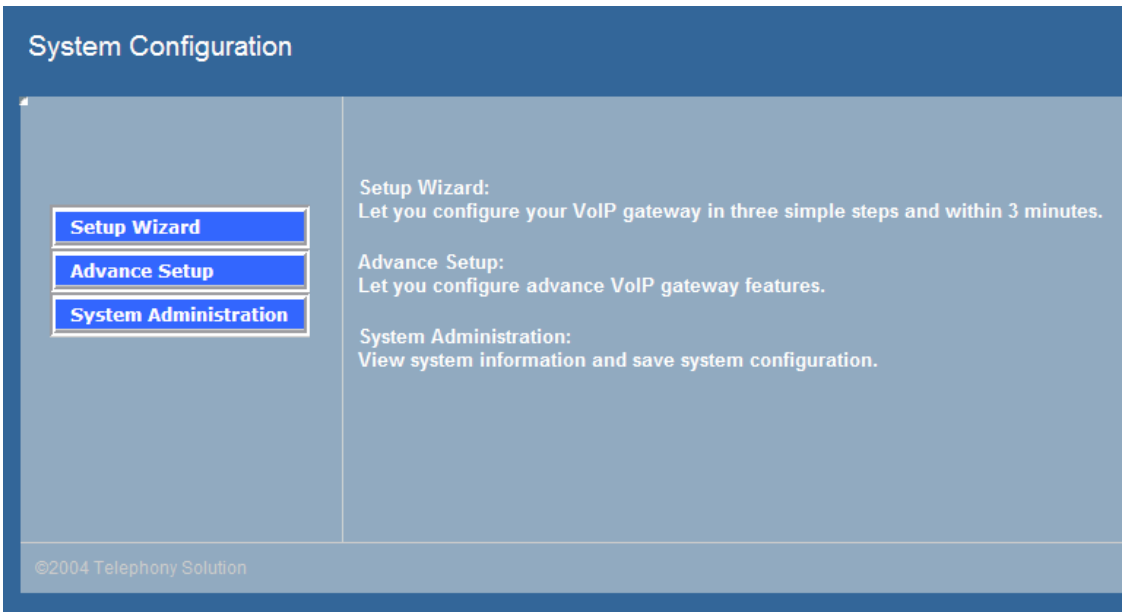
1. Launch the Web browser(IE or Firefox). Enter **http://222.222.222.1** into the browser **Address** window and press the Enter Key



2. An authentication dialog box will open.

3. If this is a first time setup of the router, type "**admin"** as the User Name and the Password field as "**admin"**. Click **OK**.(Default username/Password is "admin")

4. The Web Configuration Setup Main Menu will open. On the main page [Setup Wizard], [Advanced Setup] and [System Information] were displayed.


System Configuration
Setup Wizard
Advance Setup
System Administration
Setup Wizard:
Let you configure your VoIP gateway in three simple steps and within 3 minutes.
Advance Setup:
Let you configure advance VoIP gateway features.
System Administration:
View system information and save system configuration.
©2004 Telephony Solution

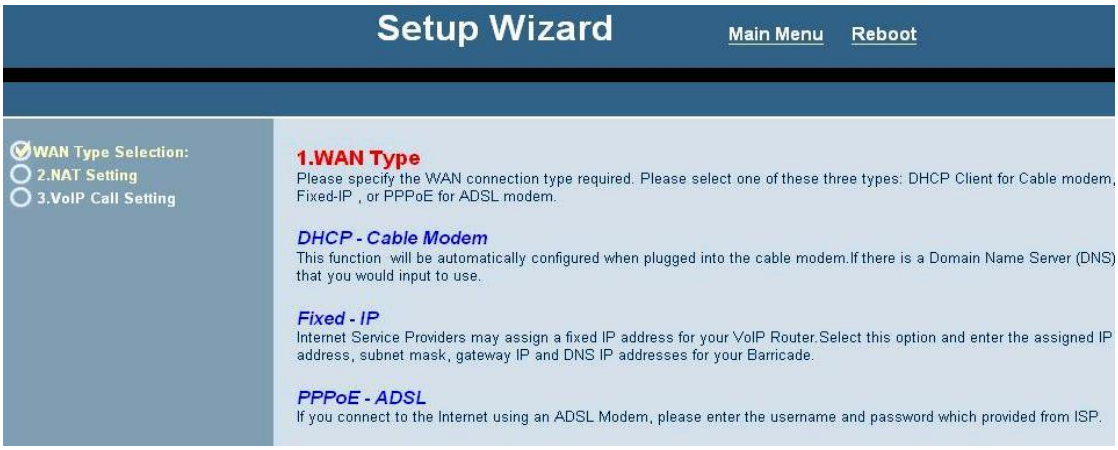
Wizard for Quick Setup

Wizard for Quick Setup gateway, After finishing the authentication, the Main menu will display 3 parts of configuration, please click “Wizard Setup” to enter quick start:

3.1 WAN Port Type Setup

For most users, Internet access is the primary application. The **S** Series Gateway support the WAN interface for Internet access and remote access. The following sections will explain more details of WAN Port Internet access and broadband access setup. When you click “**WAN Port Type Setup**” from within the Wizard **Setup**, the following setup page will be show.

Three methods are available for Internet Access:


Setup Wizard
Main Menu
Reboot
WAN Type Selection:
2.NAT Setting
3.VoIP Call Setting
1.WAN Type
Please specify the WAN connection type required. Please select one of these three types: DHCP Client for Cable modem, Fixed-IP , or PPPoE for ADSL modem.
DHCP - Cable Modem
This function will be automatically configured when plugged into the cable modem.If there is a Domain Name Server (DNS) that you would input to use.
Fixed - IP
Internet Service Providers may assign a fixed IP address for your VoIP Router.Select this option and enter the assigned IP address, subnet mask, gateway IP and DNS IP addresses for your Barricade.
PPPoE - ADSL
If you connect to the Internet using an ADSL Modem, please enter the username and password which provided from ISP.

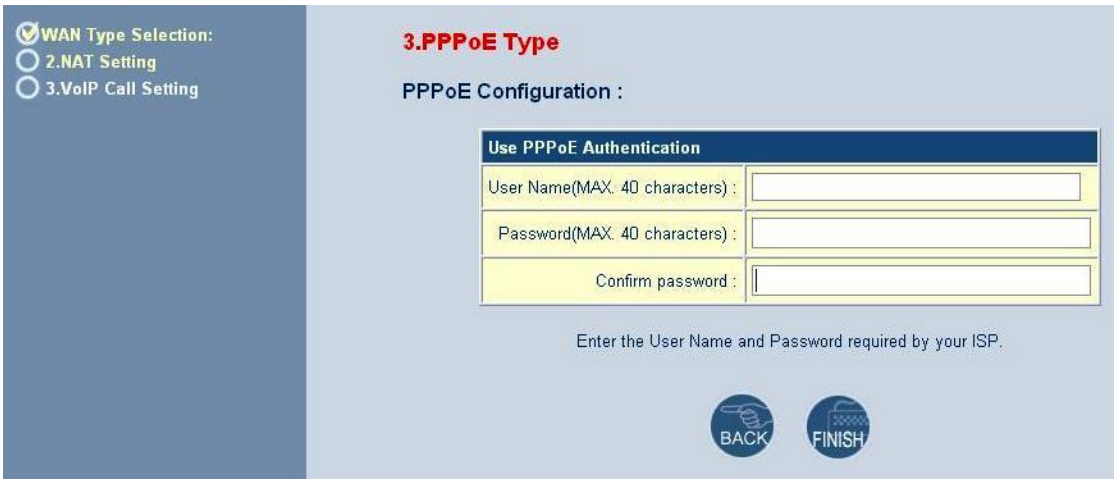
Fixed IP User: If you are a leased line user with a fixed IP address, fill out the following items with the information provided by your ISP.

Setup Wizard
Main Menu
Reboot
WAN Type Selection:
2.NAT Setting
3.VoIP Call Setting
2 Fixed-IP
IP Address
59 . 120 . 54 . 62
Default Router IP Address
59 . 120 . 54 . 254
Subnet Mask
255 . 255 . 255 . 0
Enter the IP address, Default Router IP address and Subnet Mask provided to you by your ISP in the appropriate fields above.
BACK
FINISH

- **IP Address:** check with your ISP provider
- **Netmask:** check with your ISP provider
- **Default Gateway:** check with your ISP provider

ADSL Dial-Up User (PPPoE Enable)

Some ISPs provide DSL-based service and use PPPoE to establish communication link with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to select this item.

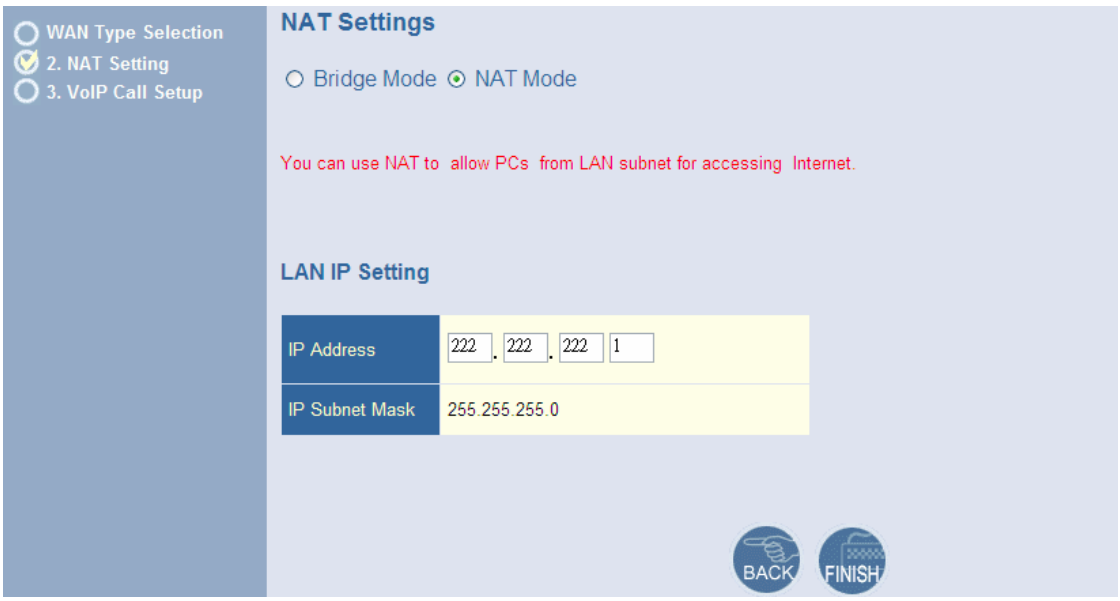


- **User Name:** Enter User Name provided by your ISP
- **Password:** Enter Password provided by your ISP.
- **Retype Password:** Enter Password to confirm again.

DHCP Client (Dynamic IP): Get WAN IP Address automatically

- **IP Address:** If you are connected to the Internet through a Cable modem line then a dynamic IP address will be assigned.

3.2 Configuring NAT or Bridge setting:

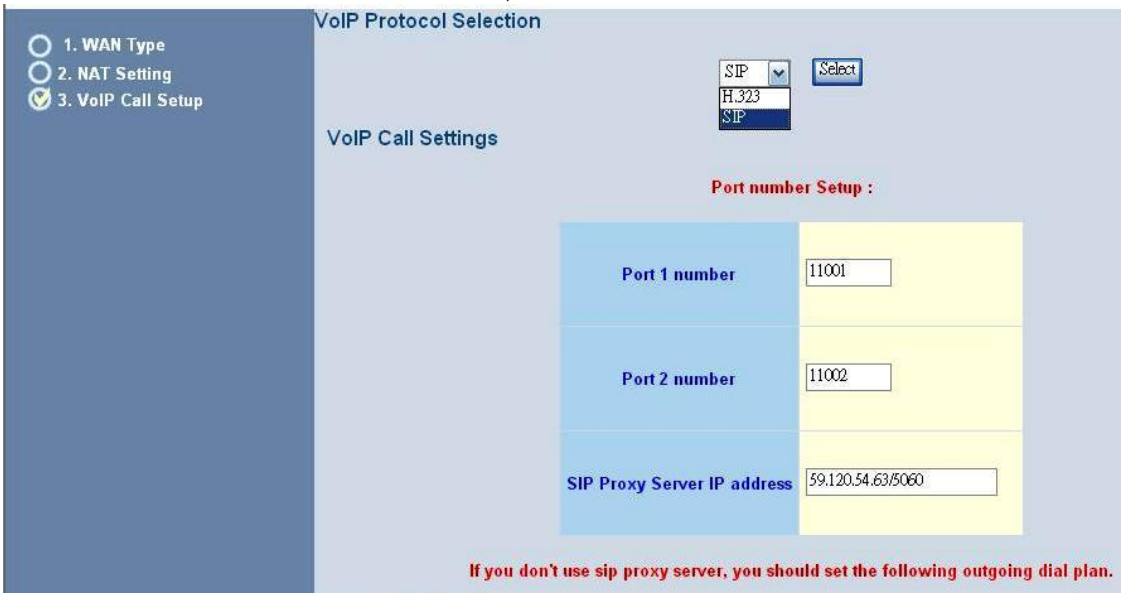


- **Bridge Mode:** Select S series Gateway as bridge.
- **NAT mode:** LAN IP Network Configuration
- **IP Address:** Private IP address for connecting to a local private network (Default: 222.222.222.51).
- **Subnet Mask:** Subnet mask for the local private network (Default: 255.255.255.0).

3.3 VoIP Call Protocol Setup

Step1 : Configure VoIP Call Signal Protocols :

- User could select H.323 or SIP Protocol, and click “select”



Step2 : configure the numbering with phone/line ports.

- **Phone Number (FXS):** The representation number is the phone number of the telephone that is connected to Phone port.

- **Line Number (FXO):** Line ports are connected to the extension ports of the PBX system or the PSTN line. They have a common Line Hunting Group Number. When this number is dialed, the Gateway system will find a free FXO line connected to PBX. This hunting will skip all busy lines and absent lines and find only the idle line to the PBX. After the available line is found, you can hear the dial tone from PBX. After that, you can dial the needed phone number out through PBX.

Step3: Let GW Register to Gatekeeper(GK)/SIP Proxy Server
(If user does not have GK/SIP Proxy Server, Please go to Step 4: Outgoing Dialing Plan)

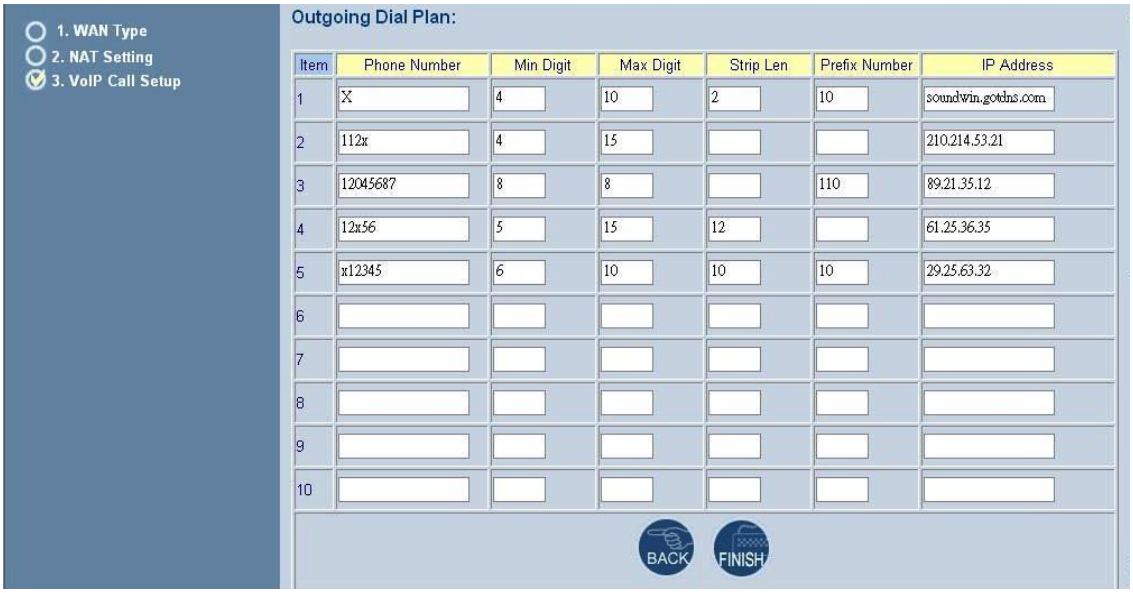
- **Gatekeeper IP address:** There is a gatekeeper address fields. If this gateway does not want to register to any gatekeeper, just set value 0.0.0.0 to the primary gatekeeper address.

- **SIP Proxy Server IP addresses:** There is a SIP Proxy Server address fields. If this gateway does not want to register to any SIP Proxy Server, just set value 0 .0.0.0 to the sip proxy server address.

Step 4: Outgoing Dialing Plan

The purpose of “Outgoing Direct Call” setting is to let user create a proprietary dialing plan when this Gateway is not registered to any H.323 Gatekeeper or any SIP Proxy Server. This setting can also assign some dialing plan to local ports (including prefix strip, prefix addition).Through this setting, user can directly map a number to a specific gateway (IP address).

In the “Outgoing Dial Plan” settings.



- “Leading Number” is the leading digits of the dialing number.
- “Min Length” and “Max Length” is the min/max allowed length you can dial.
- “Strip Length” is the number of digits that will be stripped from beginning of the dialed number.
- “Prefix Number” is the digits that will be added to the beginning of the dialed number.
- “Destination” is the IP address of the destination Gateway that owns this phone number.

Step 5: Finishing the Wizard Setup

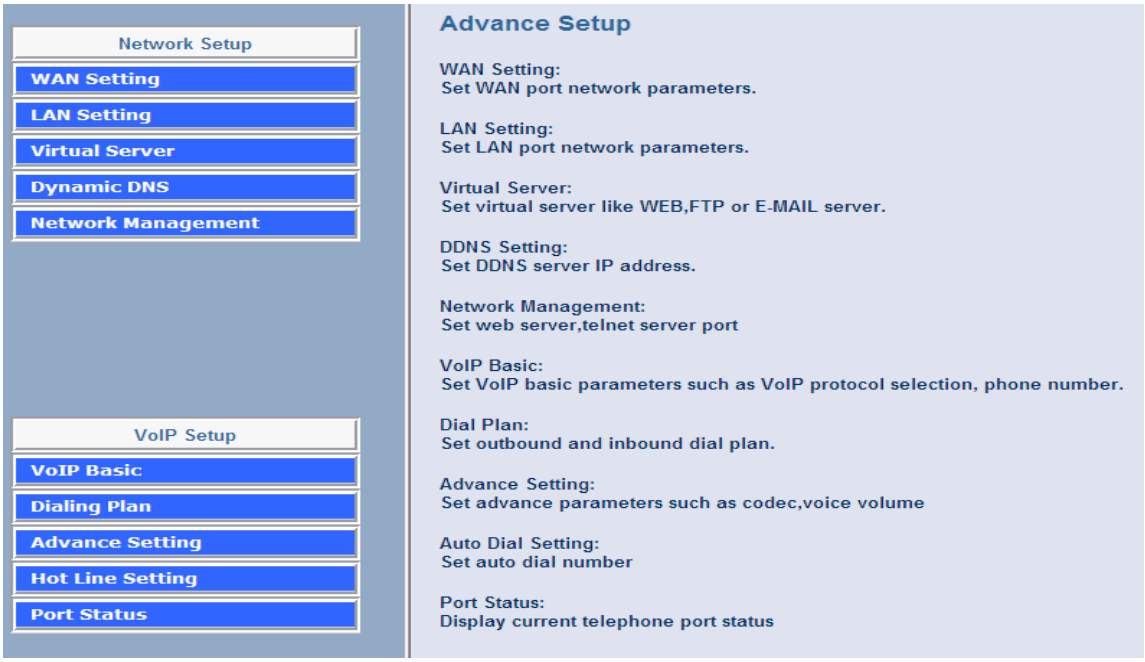
After completing the Wizard Setup, please click “Finish” bottom. The VoIP Gateway will save the configuration and rebooting Gateway automatically. After 20 Seconds, you could re-login the Gateway.

Gateway Setting

In Advanced Setup, GW provides user two major parts function to configure: One is “Network Setup”, the other one is “VoIP Call Setup”

- **SET Wizard**
- **Advance Setup**
- **System Administrator**

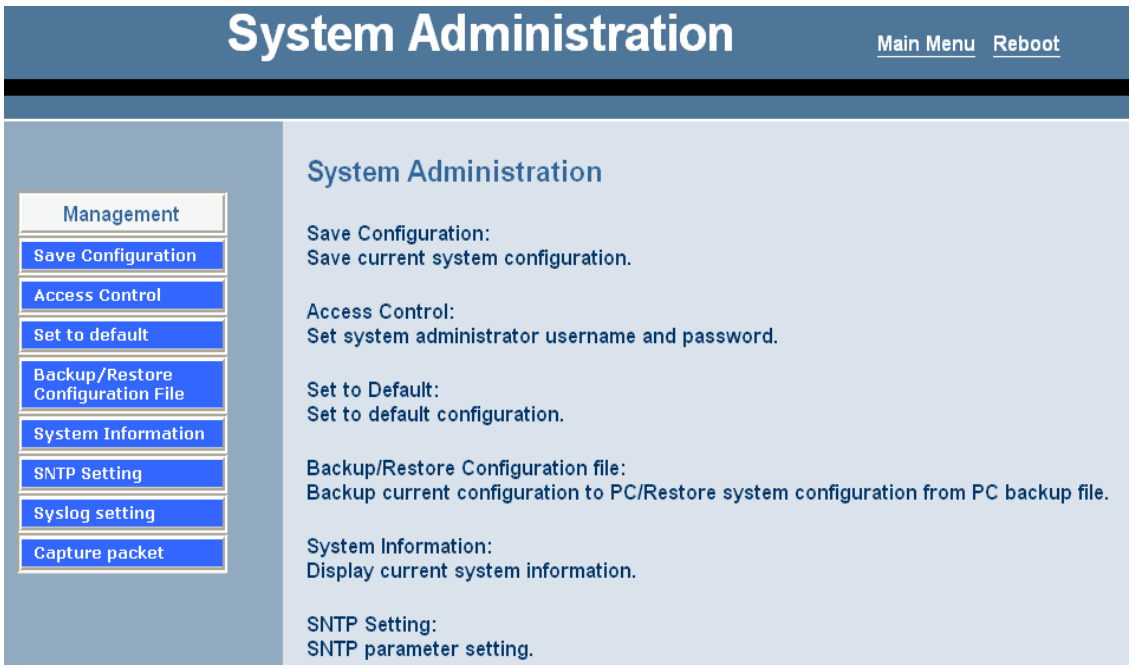
Advanced Configuration:



Advanced Setting Label

Network Setup Label	
WAN Setting	Sets/changes the WAN port Type like “Fixed IP”, “DHCP Client” or ”PPPoE”.
LAN Setting	Modifies the IP address of the LAN port and setting DHCP Server parameters.
Virtual Server	Remote user can access server such as Web or FTP at you local site via public IP address can be automatically redirected to local servers configured with private IP address.
Dynamic DNS	Dynamitic DNS allows you to provide Internet users with a domain name to access your server.
Network Parameters	Network Parameter allows you to modify the access port of gateway. For example : Setting HTTP port : 8080 Setting TELNET port is : 8081 (Default HTTP :80, TELNET: 23)
VoIP Setup	
VoIP Basic	The S Series Gateway support 2 / 4 / 8 / 16 / 24 phone/line for SIP and H.323 VoIP call applications. You can configure these ports from this menu.
Dialing Plan	Users could apply any dial policy by setting Dial Plan including outgoing dial plan and incoming dial plan.
Advanced Setting	VoIP Gateway support for silence compression, DTMF Relay, Codec Selection, FAX mode Option, H323 Register Type and H.323 Fast-Start/Normal-Start function. FXO AC impedance, Volume Adjustment, RRQ TTL, RFC2833 Payload, IP TOS,..etc
Hot Line Setting	Let user can set up “hotline” to dial the phone number automatically.
Port Status	Display the telephone interface status
Traffic Monitor	Display and Monitor the voice traffic and data traffic.

System Administration:



Management Label	
Save Configuration	You can save configuration and restart the gateway with the default configuration or with the current running configuration.
Access Control	Users can Sets/changes the administrator password..
Set to Default	You can restart the gateway with the default configuration.
Backup/Restore Configuration	User can backup the configuration file of Gateway to PC or Restore the configuration file from PC.
System Information	Display Software version, WAN Type, VoIP Status, VoIP Codec, Phone Interface and System Tim.
SNTP Setting	SNTP (Simple Network Time Protocol) Configuration for synchronizing gateway clocks in the global Internet.
Syslog Setting	Gateway can sends log information to Syslog Server by UDP ports 514.
Capture Packets	The gateway supports packets capture and save the packets to your PC. User can use Network Protocol Analyzer “Ethereal” to analysis the packets. (Free down load from http://www.ethereal.com/)

4.1 Network Configuration

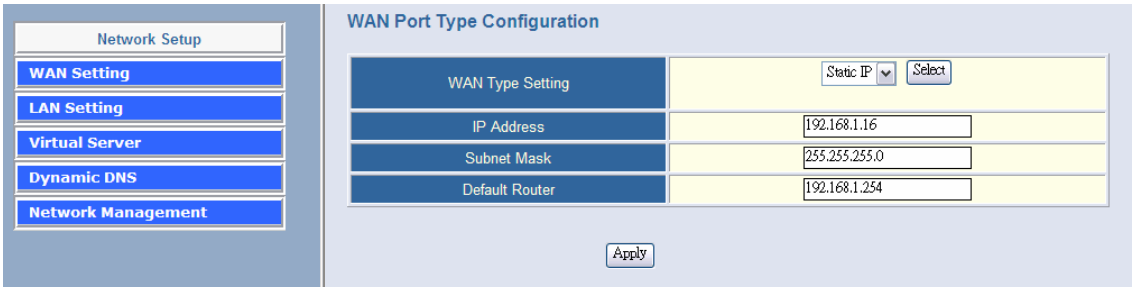
4.1.1 WAN Port Type Setup

For most users,Internet access is the primary application. The **S** Series Gateway support the WAN interface for Internet access and remote access. The following sections will explain more details of WAN Port Internet access and broadband access setup. When you click “**WAN Setting**”, the following setup page will be show. Three methods are available for Internet Access.

- **Static IP**
- **PPPoE**
- **DHCP**

Static IP:

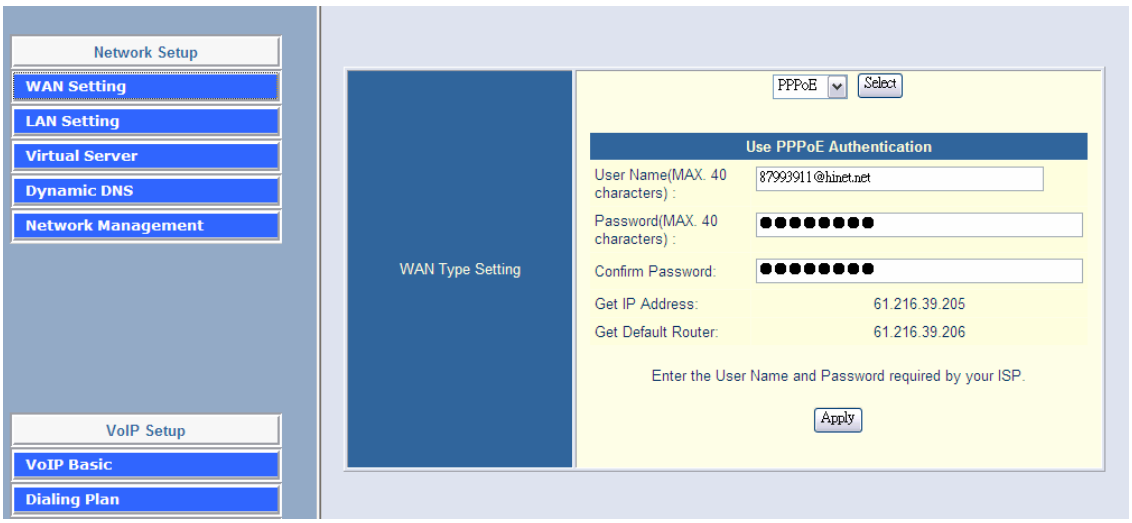
You are a leased line user with a fixed IP address; fill out the following items with the information provided by your ISP.



- **IP Address:** check with your ISP provider
- **Subnet mask:** check with your ISP provider
- **Default Gateway:** check with your ISP provider

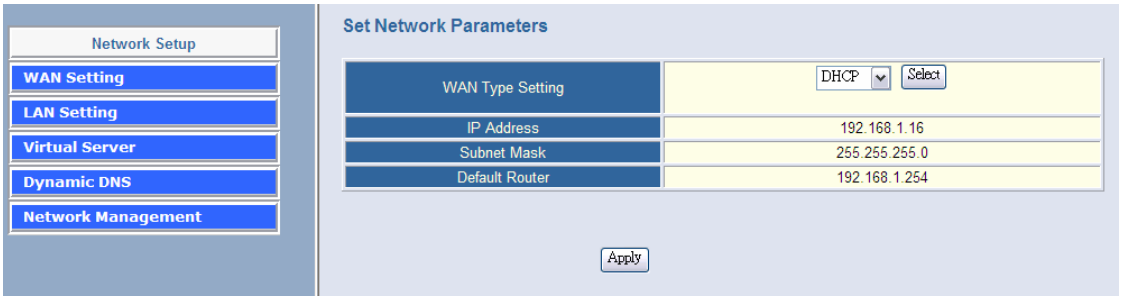
PPPoE for ADSL

Some ISPs provide DSL-based service and use PPPoE to establish communication link with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to select this item.



- **User Name:** Enter User Name provided by your ISP
- **Password:** Enter Password provided by your ISP.
- **Retype Password:** Enter Password to confirm again.

DHCP Client (Dynamic IP): Get WAN IP Address automatically

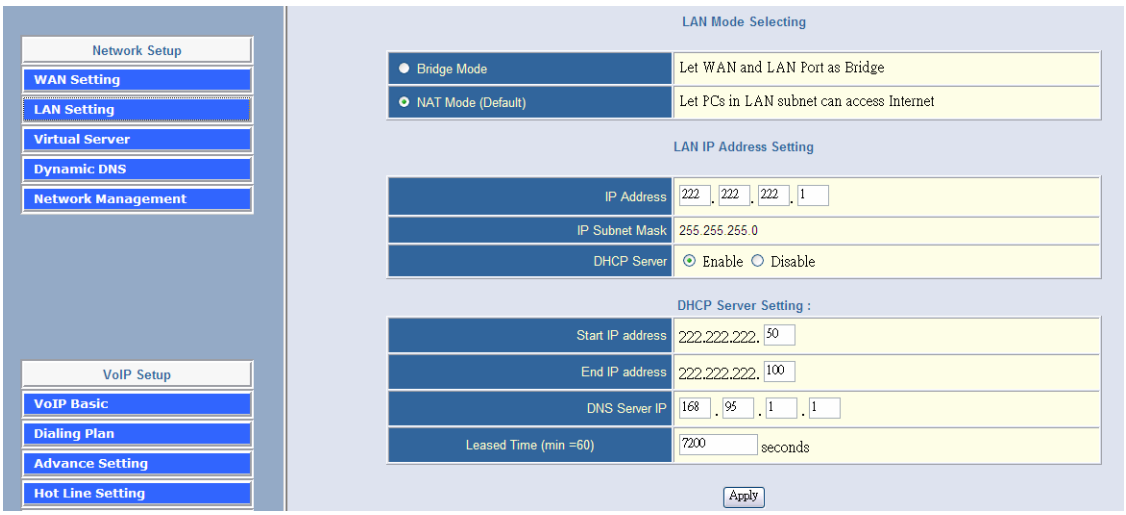


- **IP Address:** If you are connected to the Internet through a Cable modem line then a dynamic IP address will be assigned.

(Note : WAN port display the IP address, Subnet Mask and Default gateway IP address if DHCP client is successful)

4.1.2 Configuring LAN IP Address and DHCP Server

There are two kinds of network feature to configure: Bridge Mode and NAT Mode



Bridge Mode:

Select this Gateway as Bridge. Let gateway Lan port like Switch/HUB. (WAN Port and LAN Port use the same IP address)

NAT Mode:

Each of the VoIP Gateway has two Ethernet interfaces, one is for connecting to local network users, and the other is for connecting to an external broadband device (i.e. DSL modem/router or Cable modem). The LAN port is connected to the local Ethernet network. WAN is connected to the external broadband device. The LAN IP address/subnet mask is for private users or NAT users, and the WAN IP address/subnet mask is for public users.

LAN IP Network Configuration:

- **IP Address:** Private IP address for connecting to a local private network (Default: 222.222.222.1).
- **Subnet Mask:** Subnet mask for the local private network (Default: 255.255.255.0).

DHCP Server Configuration:

DHCP stands for Dynamic Host Configuration Protocol. It can automatically dispatch related IP settings to any local user configured as a DHCP client. The DHCP server supports up to 253

users (PCs) on

Yes: Enables the DHCP server. **No:** Disables the DHCP server.

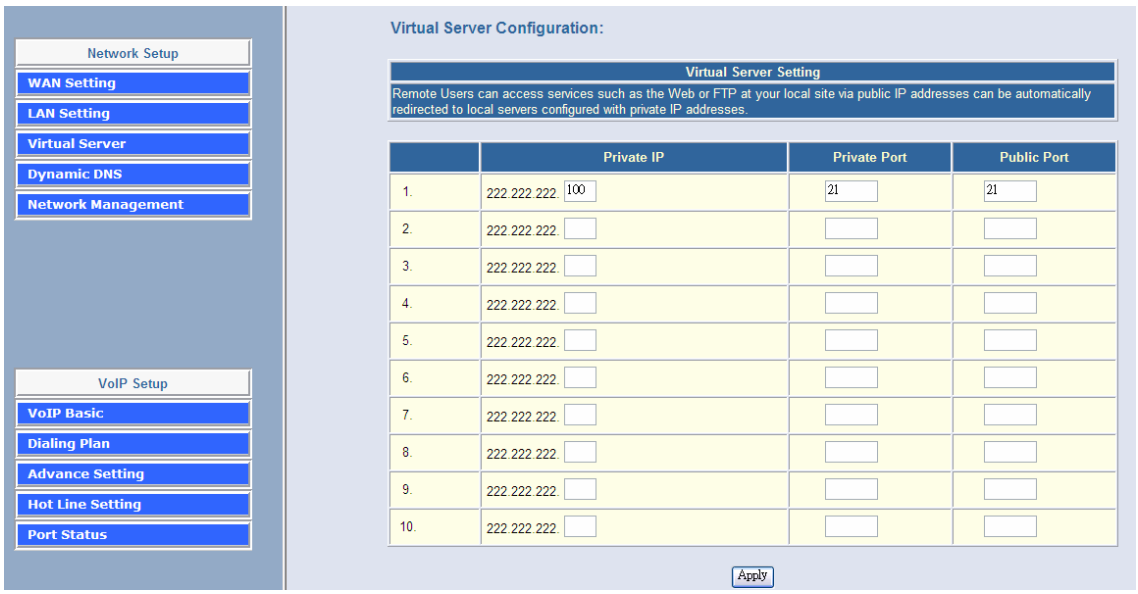
- **Start IP Address:** Sets the start IP address of the IP address pool.
- **End IP Address:** Sets the end of IP address in the IP address pool.
- **DNS Server IP Address:** DNS stands for Domain Name System. Every Internet host. must have a unique IP address, also they may have a human friendly, easy to remember name such as www.yahoo.com. The DNS server converts the human friendly name into it's equivalent IP address. (Default: None)
- **Primary IP Address:** Sets the IP address of the primary DNS server.
- **Secondary IP Address:** Sets the IP address of the secondary DNS server.

4.1.3 Virtual Server Setup

"Natural firewall" allows requests for Internet access from the local network. However, any request from the Internet to the local network is blocked. By setting the Virtual Server function, computers outside the Intranet are allowed to access specific ports of local client. The Virtual Server Port Table may be used to expose internal servers to the public domain or open a specific port number to internal hosts. Internet hosts can use the WAN IP address to access internal network services, such as FTP, WWW, Telnet etc.

How to set a Virtual Server

The following example shows how an internal FTP server is exposed to the public domain. The internal FTP server is running on the local host addressed as 222.222.222.100.



- **Private IP:** Specifies the private IP address of the internal host offering the service
- **Public Port:** Specifies which port should be redirected to the internal host..
- **Private Port:** Specifies the private port number of the service offered by the internal host.
- **Apply:** Click here to add the port-mapping entry and enable the service.

4.1.4 Dynamic DNS

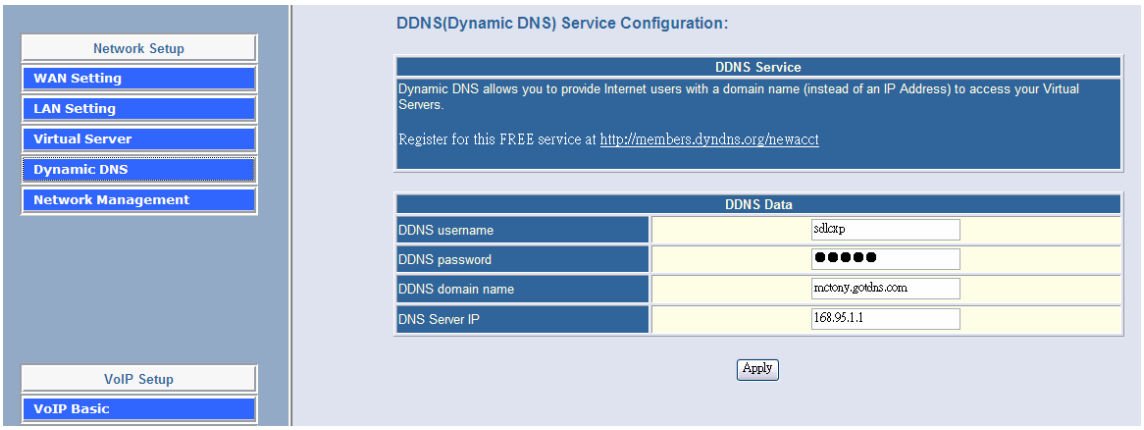
DDNS is a service that maps Internet domain names to IP addresses. DDNS serves a similar purpose to DNS: DDNS allows anyone hosting a Web or FTP server to advertise a public name to prospective users. Unlike DNS that only works with static IP addresses, DDNS works with

dynamic IP addresses, such as those assigned by an ISP or other DHCP server. DDNS is popular with home network, who typically receive dynamic, frequently-changing IP addresses from their service provider. To use DDNS, one simply signs up with a provider and installs network software on their host to monitor its IP address.

How to use DDNS

First: you should register a new DDNS service account from this web site:
http://www.dyndns.com/newacct

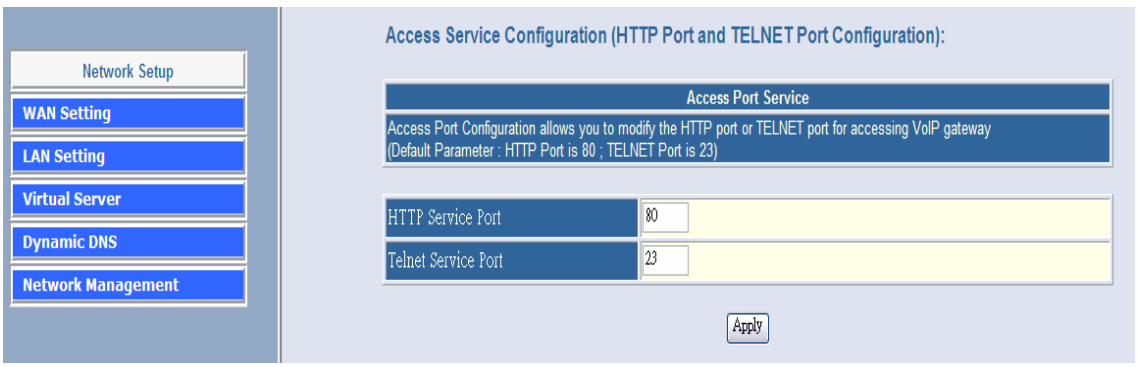
(Attention, if you use static IP address, you can't set DDNS in gateway. Use DDNS and Static IP at the same time, the dyndns will stop your DDNS service. Dyndns support DDNS service is Free, one account can create 5 different DDNS Domain Name)



- **User Name:** Input your DDNS User Name
- **Password:** Input your DDNS Password
- **Domain Name:** Input you set from your DDNS.(ie.gateway.gotdns.com)
- **DNS Server IP:** Input your DNS Server IP.

4.1.5 Network Management

Network Management,, access port configuration allows you to modify the HTTP port or TELNET port for accessing VoIP gateway
(Default Parameter : HTTP Port is 80 ; TELNET Port is 23)


Network Setup
WAN Setting
LAN Setting
Virtual Server
Dynamic DNS
Network Management
Access Service Configuration (HTTP Port and TELNET Port Configuration):
Access Port Service
Access Port Configuration allows you to modify the HTTP port or TELNET port for accessing VoIP gateway
(Default Parameter : HTTP Port is 80 ; TELNET Port is 23)
HTTP Service Port
80
Telnet Service Port
23
Apply

- **Http Server Port:** Input you want to change Web access port (Default is 80)
- **Telnet Server Port:** Input you want to change telnet access port (Default is 23)

4.2 VoIP Setup

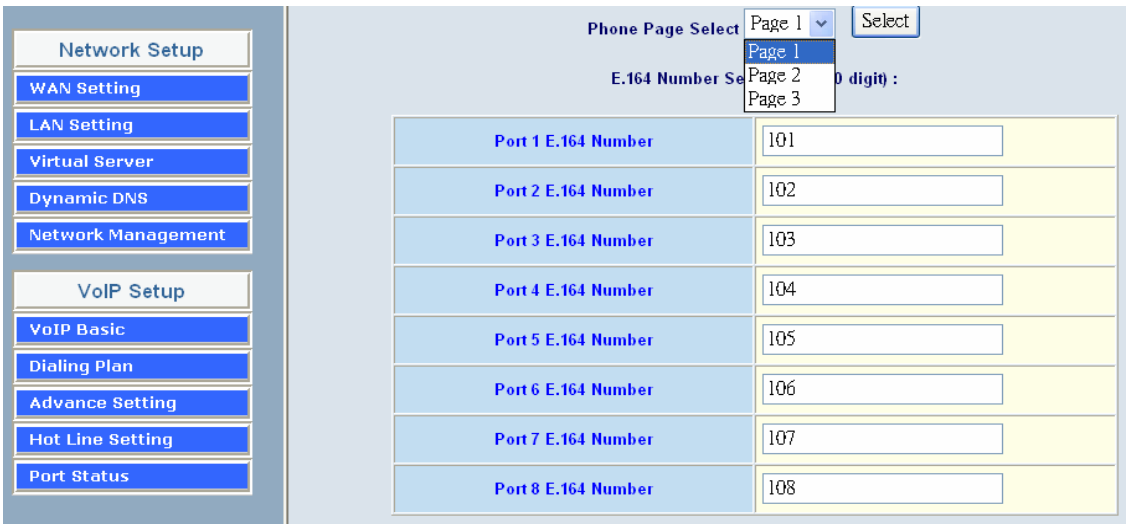
Gateway support 2 VoIP protocol - H.323 / SIP, you can register to H.323 Gatekeeper or SIP proxy server. Gateway is **not a softswitch**, it only can use 1 VoIP protocol (SIP/H.323) at the same time! If you don't register GK or Proxy server, you can make Peer to Peer call by IP address or domain name (Setting Dialing plan).

4.2.1 H.323 Setup

Gateway H.323 protocol support H.323 (v2/v3/v4), H.225, Q.931, H.245 and RTP/RTCP. Don't support **H.235 security**, can't use H.235 security Authentication Username / Password. H.323 protocol is not good at pass NAT/Firewall, the best way is installed gateway on Public IP Address when it use H.323.If you want to under NAT, gateway support NAT pass function when you use the same S Series Gateway. Other band gateway doesn't promise this function can work fine!

1. Select H.323 protocol, if you use S1600/S2400 Series, you can select different page to setting port number or others.

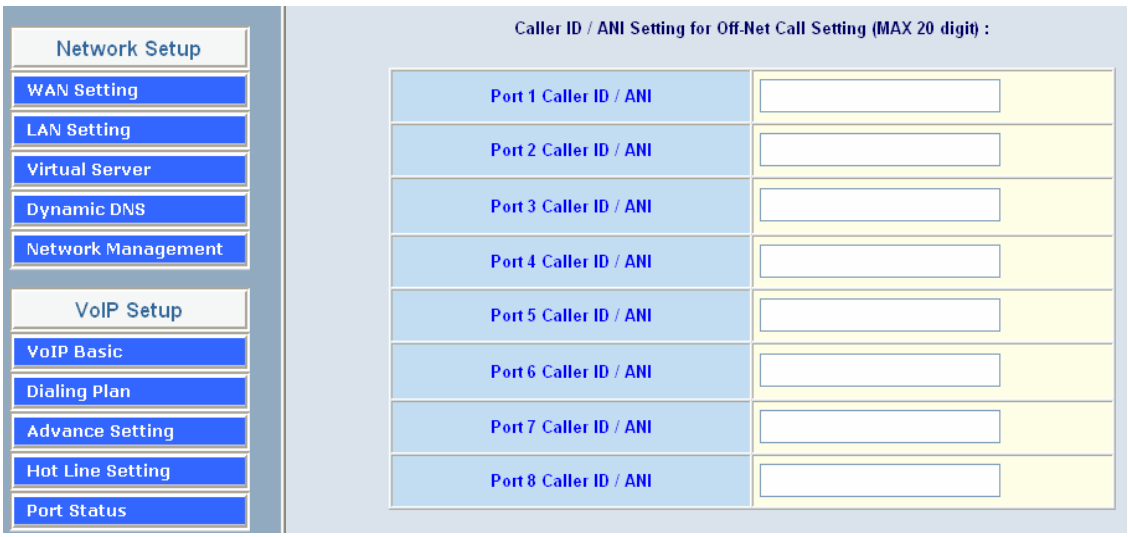
2. Configure the numbering with FXS / FXO ports.



- **FXS Number:** The representation number is the phone number of the telephone that is connected to FXS port.
- **FXO Number:** FXO ports are connected to the extension ports of the PBX system or the PSTN line. They have a common Line Hunting Group Number. When this number is dialed, the Gateway system will find a free FXO line connected to PBX. This hunting will skip all busy lines and absent lines and find only the idle line to the PBX. After the available line is found, you can hear the dial tone from PBX. After that, you can dial the needed phone number out through PBX.

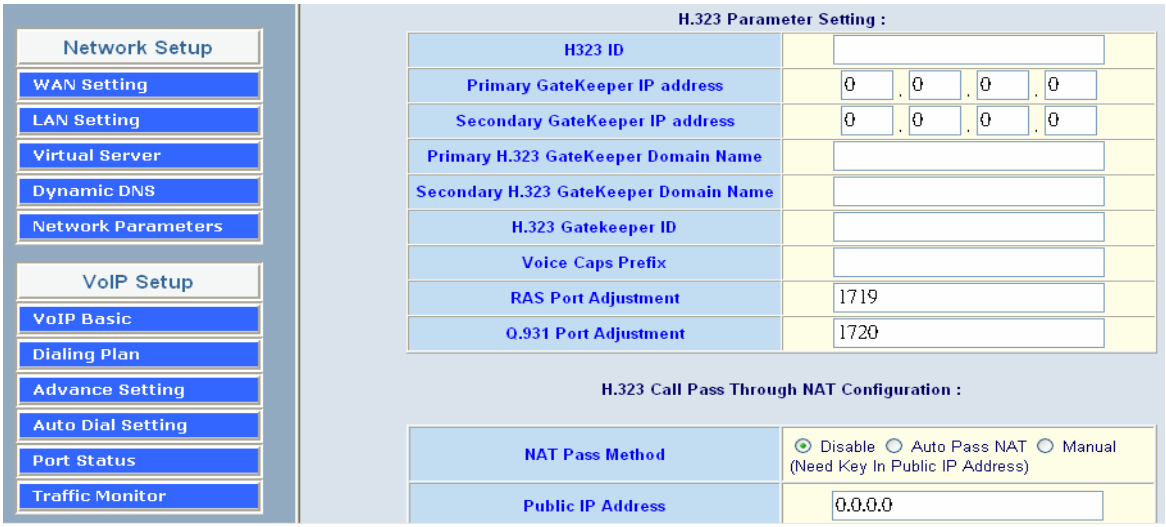
(Port number is in comparison with gateway port number. White Port socket is “FXO” port, Black Port socket is “FXS” port.)

3. Configure the ANI (Answer Number Indication) / Caller ID of the FXS/FXO ports.


Network Setup
WAN Setting
LAN Setting
Virtual Server
Dynamic DNS
Network Management
VoIP Setup
VoIP Basic
Dialing Plan
Advance Setting
Hot Line Setting
Port Status
Caller ID / ANI Setting for Off-Net Call Setting (MAX 20 digit) :
Port 1 Caller ID / ANI
Port 2 Caller ID / ANI
Port 3 Caller ID / ANI
Port 4 Caller ID / ANI
Port 5 Caller ID / ANI
Port 6 Caller ID / ANI
Port 7 Caller ID / ANI
Port 8 Caller ID / ANI

- ITSP needs ANI for authorization when gateway calls Off-Net call to PSTN number or mobile phone number.

4. Register to H.323 Gatekeeper
(If user does not have Gatekeeper, Please go to Dialing Plan Policy)

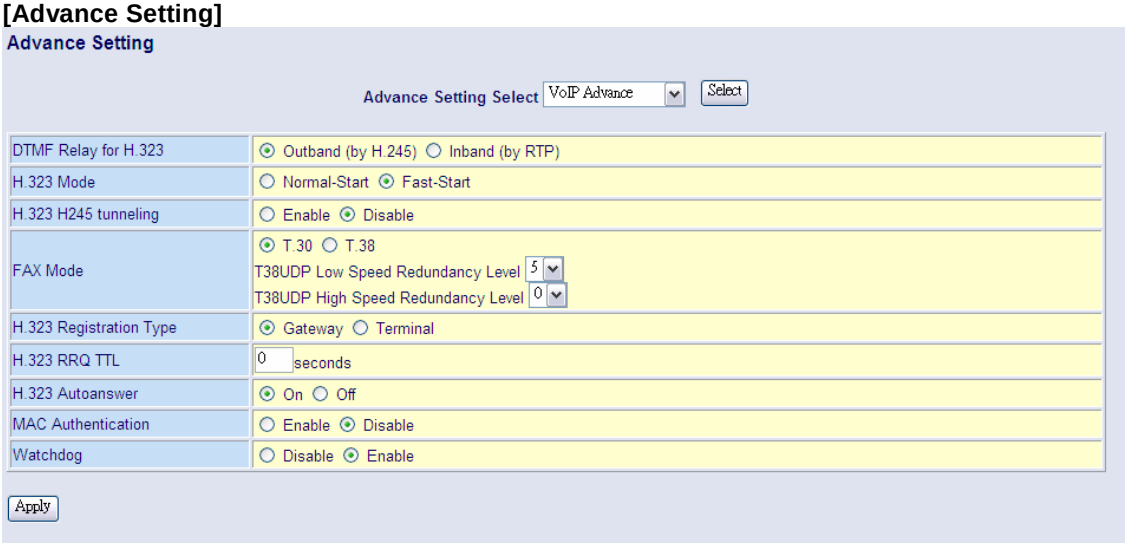


H.323 Parameters Label	
H.323 ID	Sets the unique name of this Gateway, that is communicated as part of H.323 messaging..
Primary Gatekeeper IP Address Secondary Gatekeeper IP Address	There are two gatekeeper address fields, one is primary, the other secondary. If this gateway does not want to register to any gatekeeper, just set value 0 to the primary gatekeeper address. If the primary gatekeeper address is not 0, the gateway will register to the primary gatekeeper. If the second gatekeeper is not 0, the gateway will try to register to the second gatekeeper when failed to register to primary gatekeeper, i.e. if both the primary gatekeeper and second gatekeeper addresses are present, the gateway will try to register to these two gatekeepers respectively. The gateway can have the gatekeeper backup function by this way.
Primary Gatekeeper Domain Name Secondary Gatekeeper Domain Name	Let user use Domain Name of H.323 Gatekeeper.
H.323 Gatekeeper ID	The Gatekeeper ID; usually do not need to set this field unless the gatekeeper must need this value.
Voice Cap Prefix	Let user set prefix number in RRQ nonstandard voice cap entry.
RAS Port Adjustment	In H.323 standard the RAS default port number is 1719. The VoIP gateway provides user to change RAS port number to meet the network environment.(Some area carrier blocks or forbidden the default port number)
Q.931 Port Adjustment	In H.323 standard the default Q.931 port number is 1720. The VoIP gateway provides user to change Q.931 port to meet the network environment. (Some area carrier blocks or forbidden the default port number)
H.323 Call Pass through NAT	
H.323 Pass Through NAT method	1. Disable : The Gateway operates in public IP address 2. Auto Detection: When the Gateway register to GNU Gatekeeper / H.323 Gatekeeper (SK Series), please select this option. 3. Manual Setting: When the Gateway registers to H.323 Gatekeeper and operate under NAT (enable DMZ), please select this option and key in IP address.

H.323 VoIP Advanced Configuration

There are many H.323, VoIP, Codec and other more detail Setting, you can set in "**Advance Setting**". For SIP and H.323, there are a little different in advance setting. There are 3 different parts to setting about VoIP, Telephone and network.

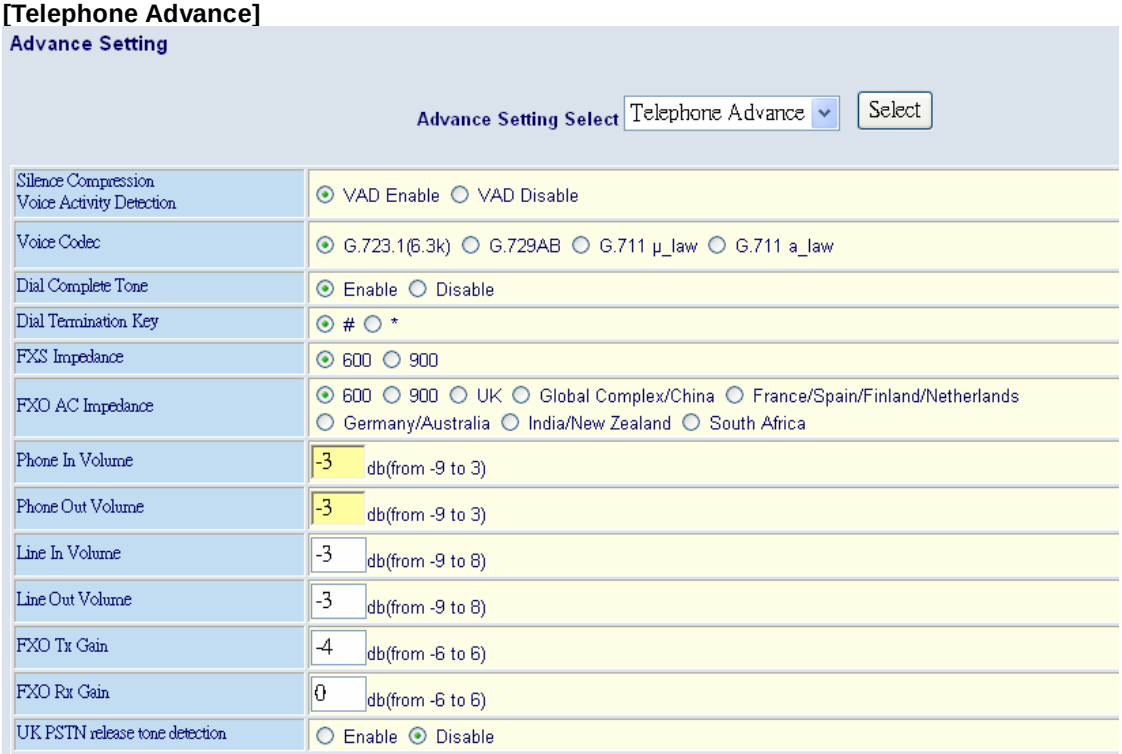
[Advance Setting]

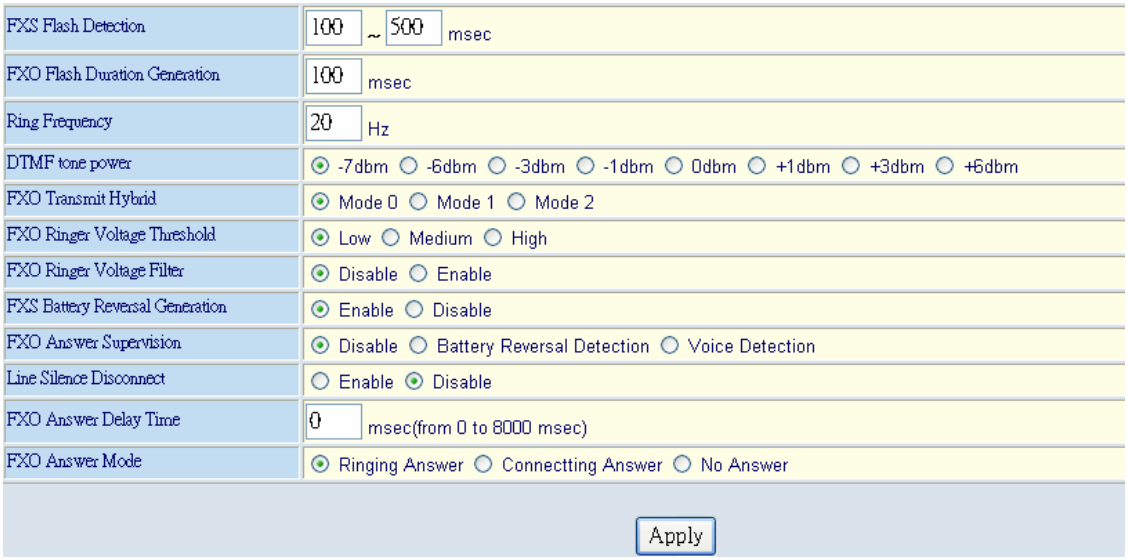


Item	Description
DTMF Relay for H.323:	After the VoIP call is connected, when you dial a digit, this digit is sent to the other side by DTMF tone. There are two methods of sending the DTMF tone. The first is "in band", that is, sending the DTMF tone in the voice packet. The other is "out band", that is, sending the DTMF tone as a signal. Sending DTMF tone as a signal could tolerate more packet loss caused by the network. If this selection is enabled, the DTMF tone will be sent as a signal.
H.323 Mode:	This selection could force the Gateway to use normal start mode (default mode) or fast start mode when establishing a VoIP call. Many other gateways only support normal start mode, enable this selection when it is necessary. The default is disabled (using fast start mode).
H.323 H.245 Tunneling:	This selection could force the Gateway to use H.245 Tunneling when establishing a VoIP call The default is disabled (using fast start mode).
FAX Mode Option:	T.30/T.38 real-time FAX compliant Voice/FAX auto-switch. The T.38 is a "Real Time Group 3 Fax Communication over IP network" format. That's meaning it's a protocol for Fax over IP. You have to enable this function (T.38 mode isn't support all gateway, different band use T.38 have a little change, it maybe let T.38 FAX Error)
H.323 Registration type:	There are 2 choices for this setting. "Gateway" means it will act as the VoIP gateway. "Terminal" means it will act as the IP phone terminal.
H.323 RRQ TTL:	This command configures the number of seconds that the gateway should be considered active by the H.323 gatekeeper. The gateway transmits this value in the RRQ message to the gatekeeper. The default value is "0".
H.323 Autoanswer:	When a VoIP call is incoming, the Gateway will ring a specific

	phone set. The H.323 call signaling part could be connected or alerting during this ringing period. If this selection is enabled, the H.323 signaling part is connected during the ringing period. The benefit of this situation is that the remote side could hear the status of the specific port. That is, the remote side will hear ring back tone if the Gateway is really ringing the phone set. If the phone set is busy, the remote side will hear busy tone. The disadvantage of this situation is that the H.323 connected time is not the real voice call connected time. So, if billing is recorded for this Gateway, this function should be disabled.
MAC Authentication:	Some Gatekeeper register need UA send MAC address to Authentication, you need enable this function.(Default is disable).
Watchdog:	When your gateway shutdown, or something happen that made gateway can't work fine. Watchdog will reboot your gateway automatically when it can't work.

[Telephone Advance]

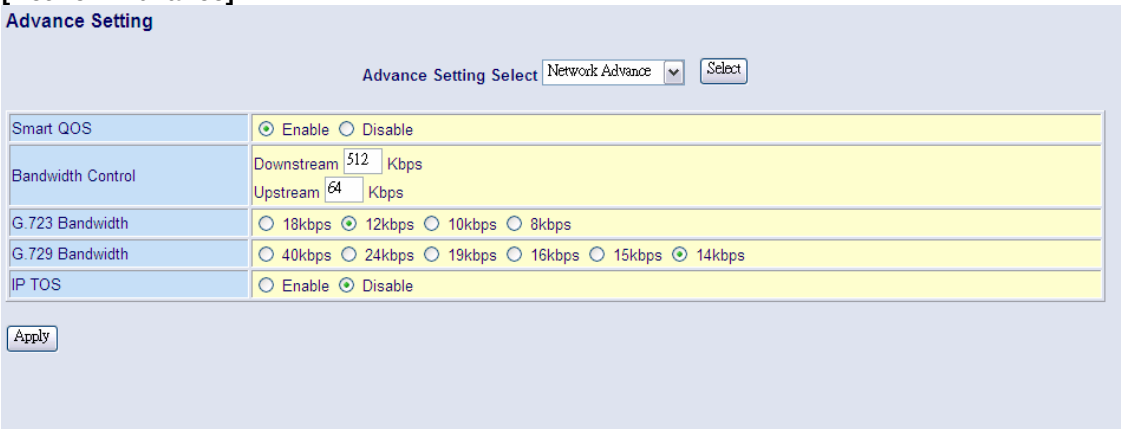




Item	Description
Silence Compression: (VAD)	If this function is enabled, when silence is occurred for a period of time, no data will be sent across the network during this period in order to save bandwidth. (If you use Asterisk, please disable Silence Compression, it maybe make you call disconnect.)
Voice Codec option:	The Codec is used to compress the voice signal into data packets. Each Codec has different bandwidth requirement. There are four kinds of Codec, G.723, G.729AB, G.711_u and G.711_A. The default value is G.723.
Dial Complete Tone:	When you use the VoIP call, you will heard “DuDu” voice that is dial complete tone. If you don’t want to heard that tone , you can disable it.(default is enable).
Dial Termination key:	Setting Termination key to speed up VoIP dial. Select “*” or “#” to Termination key.
FXS Impedance:	The FXS provides 600/900 OHM impedances for selection.
FXO AC Impedance:	The FXO provides wild and complex ac termination impedances for selection.
Phone (Line) in/out volume:	You can adjust the Phone (Line) in/out volume, range from -9db to 9db (If you adjust too bigger, maybe generation some ECHO or noise)
FXO Tx/Rx Gain:	You can adjust the FXO Tx/Rx Gain , range from -6db to 6db. (If you adjust too bigger, maybe generation some ECHO or noise)
UK PSTN release tone detection:	When you use the Gateway to UK, you can Enable this selection to detection release tone
FXO/FXS Flash detection and generation duration :	Flash Detection: Let you change flash detection (milliseconds) of Gateway when phone generate flash to FXS. Flash Generation: Let you change flash generation time (milliseconds) for PBX detection.
Ring Frequency:	You can configure how long the Ring Frequency do you want to use.
DTMF tone power:	Sometimes you input DTMF, but no request. You can adjust this function, range from -6db to +6db.
FXO Transmit Hybrid:	ECHO cancellation adjust, 3 mode setting to solve ECHO problem, default is Mode 1. If you have echo problem, try to select

	other Option.
FXO Ringer Voltage Threshold:	Sometime you use FXO connect to PSTN/PBX can't be pick up the call. PSTN/PBX can't detection FXO voltage, you can adjust this function. Default is low.(Low : 16.5 Vrms Medium: 24 Vrms High: 49.5 Vrms)
FXO Ringer Voltage filter:	Some vendor's PBX generates the leakage voltage from extension port. That will mislead the FXO become Off-hook status. This function was set to avoiding a leakage voltage signal is detected as ring coming.
FXS Battery Reversal Generation:	Some Payphone need Battery Reversal to count call money. Gateway FXS can generation that Battery Reversal. If you don't use this function , don't enable it.
FXO Answer Supervision	Enable battery reverse to detect polarity from PSTN line. The PSTN line can send. H.323 case: Sending the Q.931 connect signal to caller when detecting polarity reverse from PSTN Line. SIP case: Sending the 200 OK connect signal to caller when detecting polarity reverse from PSTN Line. Voice Detection: when the FXO connect PSTN/PBX, gateway will detection the PSTN/PBX send voice , it will send Q.931 connect or SIP 200 ok.
Line Silence Disconnect:	When FXO call have silence, the call will disconnect automatically. {Default is Disable}
FXO Answer Delay Time	Some PSTN/PBA when call incoming can't pick up first time. It need delay few second time. You can set FXO pick up call delay time.(default is 0)
FXO Answer Mode	FXO Answer Mode Concept: When user calls the PSTN line which was connected with the FXO port, there are three answer mode for user to configure. 1. Ringing Answer Mode (Default Setting): FXO answer the call once the ring coming from PSTN line. 2. Connecting Answer Mode: **Case A:** "Hot Line Number" was NOT assigned in the FXO port. FXO answer the call once the ring come from PSTN line. **Case B:** "Hot Line Number" was assigned and the Hot line number belongs to remote VoIP device. In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call. (Note: This case can avoid charging for the Local PSTN call when the remoting VoIP devic still ring.) **Case C:** "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway. FXO port will not answer (off-hook) till the Phone (connected to the FXS port) was picked up by user. (Note: This case can avoid the Local PSTN charge when the FXS port still ring.) 3. Non Answer Mode: FXO will NOT answer the call in any time. (Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination)

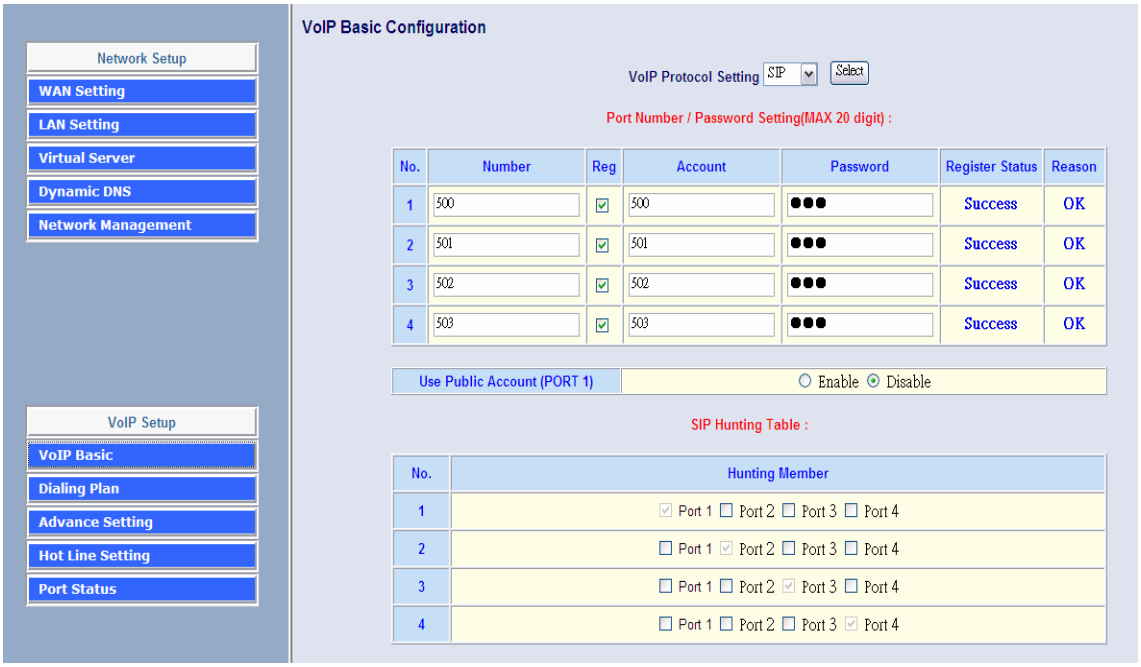
[Network Advance]



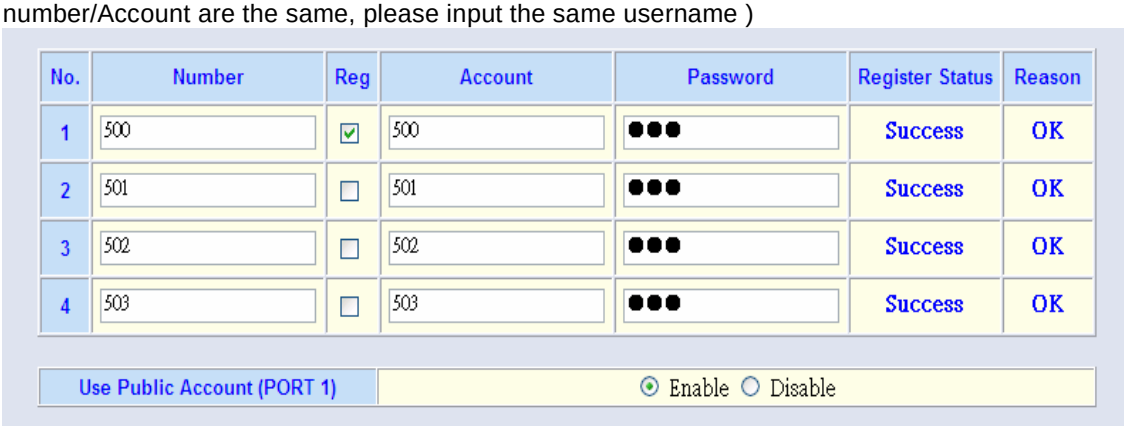
Item	Description
Smart-QoS:	If this function is enabled, when VoIP call is occurred, the other data will be automatically reduced traffic which across the internet in order to guarantee the voice bandwidth.
Bandwidth control:	You can configure your bandwidth what the Max byte of download and upload of ADSL modem rate.
G.723/G.729 Bandwidth:	Setting G.723 / G.729 voice compression size. Quality and Packet size can adjust by you want.
IP TOS:	Some Router support TOS(Type of Service), when you enable the TOS function, the router will process those packets firstly.(default is disable)

4.2.2 SIP Setup

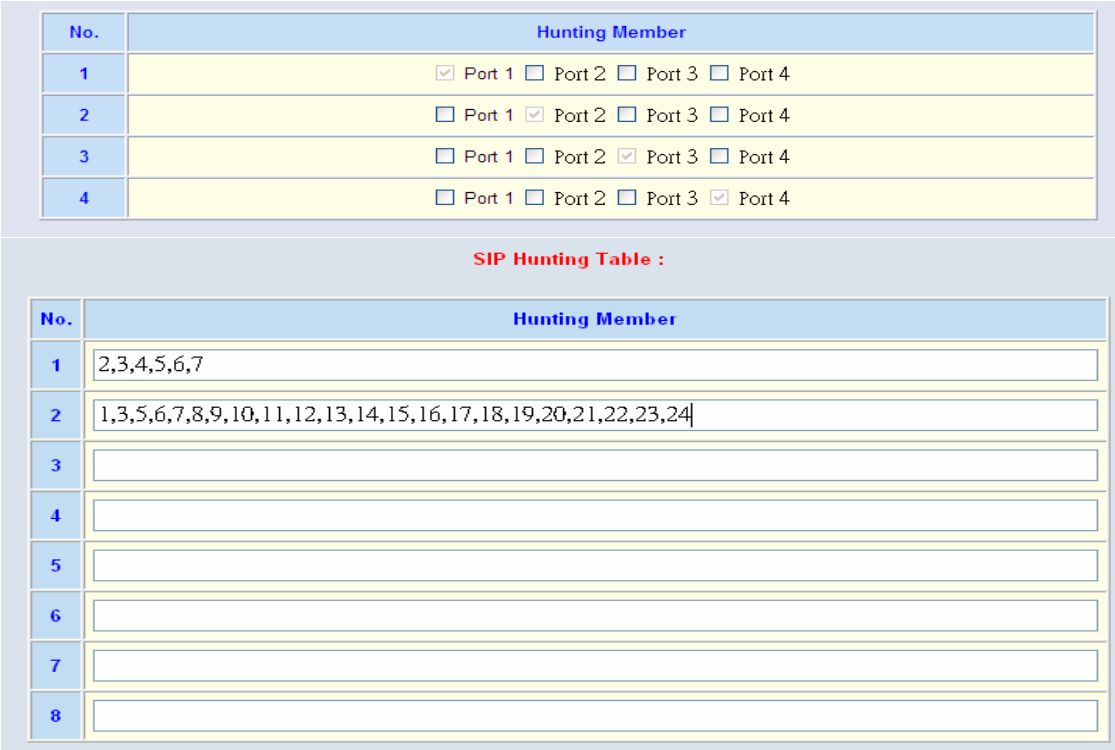
Gateway SIP support SIP(RFC3261), SDP(RFC2327), RFC2833, STUN(RFC3489), Symmetric RTP, outbound proxy, ENUM(RFC2916),and RTP/RTCP.SIP NAT pass through Function can support 80% NAT/Firewall that you don't setting DMZ/Virtual server in router or Firewall.


Network Setup
WAN Setting
LAN Setting
Virtual Server
Dynamic DNS
Network Management
VoIP Setup
VoIP Basic
Dialing Plan
Advance Setting
Hot Line Setting
Port Status
VoIP Basic Configuration
VoIP Protocol Setting SIP Select
Port Number / Password Setting(MAX 20 digit) :
No. Number Reg Account Password Register Status Reason
1 500 500 Success OK
2 501 501 Success OK
3 502 502 Success OK
4 503 503 Success OK
Use Public Account (PORT 1) Enable Disable
SIP Hunting Table :
No. Hunting Member
1 Port 1 Port 2 Port 3 Port 4
2 Port 1 Port 2 Port 3 Port 4
3 Port 1 Port 2 Port 3 Port 4
4 Port 1 Port 2 Port 3 Port 4

1. Select "SIP Protocol"
2. SIP number / account (username) and Password Setting: Please fill out the SIP account including username / password from ITSP.
(Note: support digits and character base SIP Account / username, some SIP Server use character username to login, and a number to call number(ie. VoIPBuster) , if your server don't support this, number/Account are the same, please input the same username)

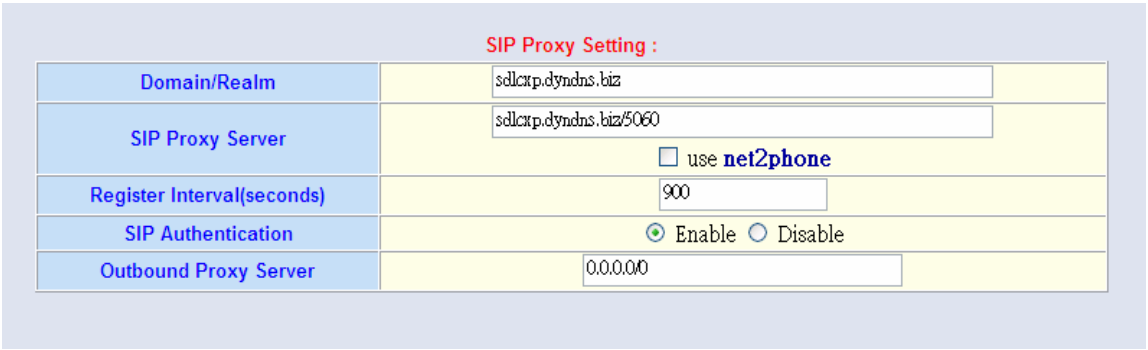

No. Number Reg Account Password Register Status Reason
1 500 500 Success OK
2 501 501 Success OK
3 502 502 Success OK
4 503 503 Success OK
Use Public Account (PORT 1) Enable Disable

- **Number:** Input SIP Number(Username), if your server support account and number (different),input the number, else number/account are the same username.
- **Reg:** let your sip account register SIP Server, click this option.
- **Account:** Input SIP account(Username), if your server support account and number (different),input the number, else number/account are the same username.
- **Password:** Input Password that ITSP support.
- **Use Public Account:** This allows gateway can use single SIP account for multiple ports. User input the only one account in port one field for registering the ITSP.



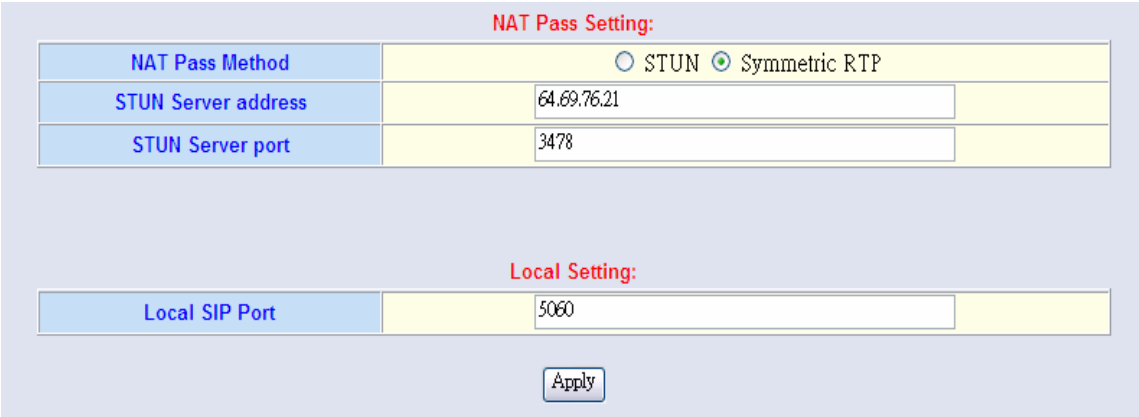
- **SIP Hunting Table:** This allows gateway can answer SIP call from internet by Hunting. For example: Port 2-7 are hunting for the Port 1 SIP account. When port 1 are on call, the other one SIP call from internet will ring port 2.S200/400/800 Series hunting table is use click box.

3. SIP Proxy Server setting, setting SIP proxy server register information.
(If user does not need register SIP Proxy Server, Please go to Dialing Plan Policy)



SIP Proxy Server Label	
SIP Proxy Server Setting	1. Enter the SIP service IP address or domain name in this field (the domain name that comes after the @ symbol i n a full SIP URI). 2. Use Net2Phone Service Provider
SIP Domain	1. Enter the SIP realm in this field
Register Interval Setting	This field sets how long an entry remains registered with the SIP register server. The register server can use a different time period. The Gateway sends another registration request after half of this configured time period has expired.
SIP Authentication	Enable or Disable MD5 Authentication with SIP Proxy Server

4. If your gateway under the NAT/Firewall, you should setting different NAT Pass function. if you setting STUN/Outbound Proxy, you should have a STUN/Outbound proxy server. If they can't pass NAT or one way talk happen, try to open "DMZ" and virtual server "5060" port in router.



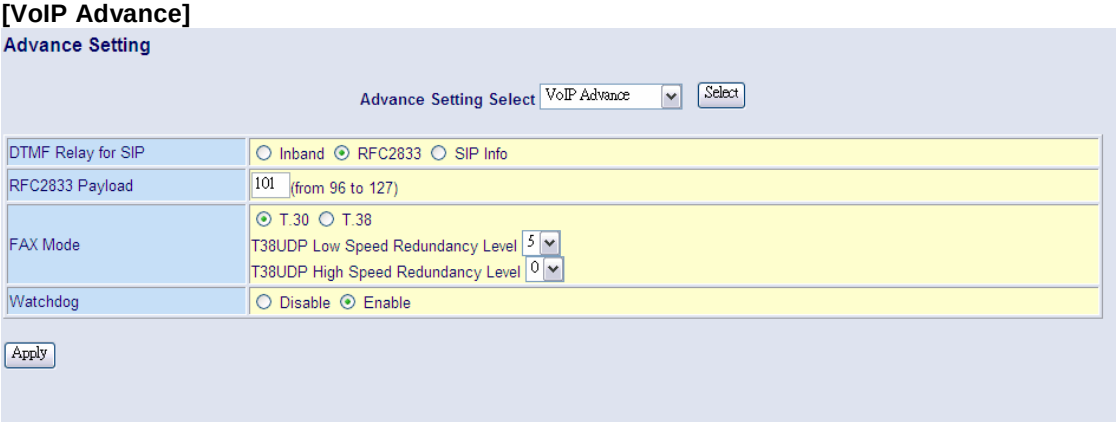
- **Symmetric RTP:** default use Nat pass function.
- **STUN Client:** setting your STUN server information, default STUN server is FWD STUN server.
- **Outbound Proxy Support:** Setting your Outbound Proxy server information.

- **Local SIP Port::** setting local use SIP port, default is 5060.

SIP VoIP Advanced Configuration

There are many SIP VoIP, Codec and other more detail Setting, you can set in "**Advance Setting**". For SIP and H.323, there are a little different in advance setting. There are 3 different parts to setting about VoIP, Telephone and network.

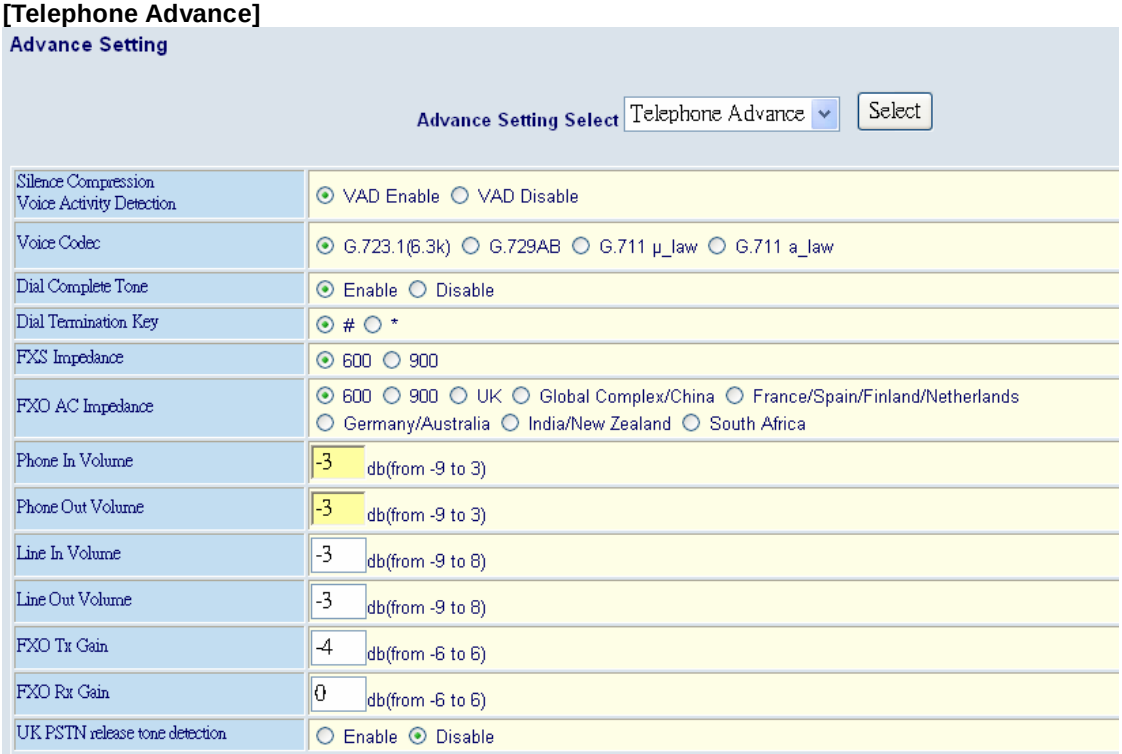
[VoIP Advance]

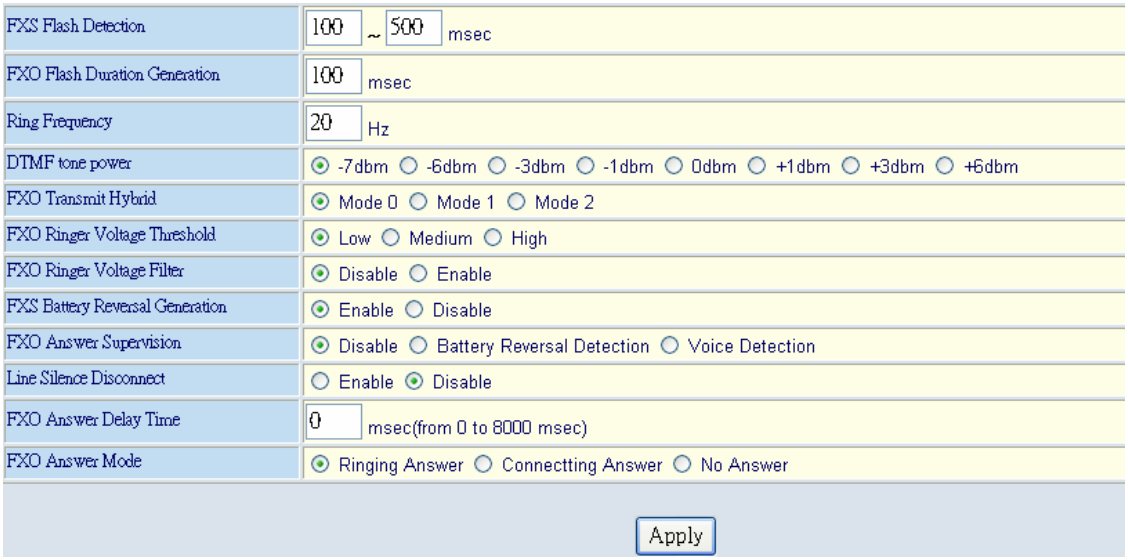


Item	Description
DTMF Relay for SIP:	After the VoIP call is connected, when you dial a digit, this digit is sent to the other side by DTMF tone. There are three methods of sending the DTMF tone. The first one is "in band", that is, sending the DTMF tone in the voice packet. The second one is "RFC2833", that is, sending the DTMF tone as a RTP payload signal. The third one is "SIP Info", that is, sending the DTMF tone as a SIP signal. Sending DTMF tone as a signal could tolerate more packet loss caused by the network. If this selection is enabled, the DTMF tone will be sent as a

	signal.
RFC2833 Payload:	Adjust RFC2833 DTMF payload value, range from 96 to 127, default is 101.
FAX Mode Option:	T.30/T.38 real-time FAX compliant Voice/FAX auto-switch. The T.38 is a “Real Time Group 3 Fax Communication over IP network” format. That’s meaning it’s a protocol for Fax over IP. You have to enable this function (T.38 mode isn’t support all gateway, different band use T.38 have a little change, it maybe let T.38 FAX Error)
Watchdog:	When your gateway shutdown, or something happen that made gateway can’t work fine. Watchdog will reboot your gateway automatically when it can’t work.

[Telephone Advance]

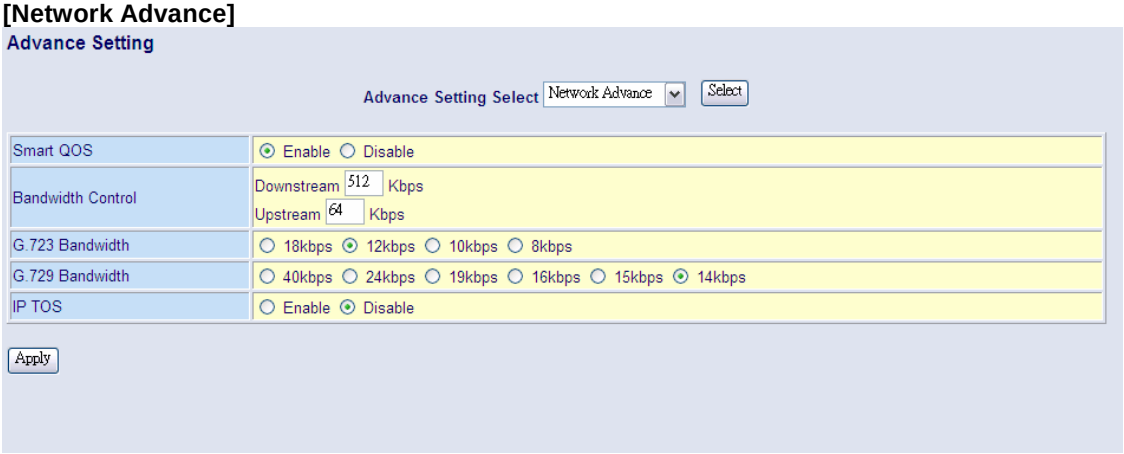




Item	Description
Silence Compression: (VAD)	If this function is enabled, when silence is occurred for a period of time, no data will be sent across the network during this period in order to save bandwidth. (If you use Asterisk, please disable Silence Compression, it maybe make you call disconnect.)
Voice Codec option:	The Codec is used to compress the voice signal into data packets. Each Codec has different bandwidth requirement. There are four kinds of Codec, G.723, G.729AB, G.711_u and G.711_A. The default value is G.723.
Dial Complete Tone:	When you use the VoIP call, you will heard “DuDu” voice that is dial complete tone. If you don’t want to heard that tone , you can disable it.(default is enable).
Dial Termination key:	Setting Termination key to speed up VoIP dial. Select “*” or “#” to Termination key.
FXS Impedance:	The FXS provides 600/900 OHM impedances for selection.
FXO AC Impedance:	The FXO provides wild and complex ac termination impedances for selection.
Phone (Line) in/out volume:	You can adjust the Phone (Line) in/out volume, range from -9db to 9db. (If you adjust too bigger, maybe generation some ECHO or noise)
FXO Tx/Rx Gain:	You can adjust the FXO Tx/Rx Gain , range from -6db to 6db. (If you adjust too bigger, maybe generation some ECHO or noise)
UK PSTN release tone detection:	When you use the Gateway to UK, you can Enable this selection to detection release tone
FXS/FXO Flash detection and generation duration :	Flash Detection: Let you change flash detection (milliseconds) of Gateway when phone generate flash to FXS. Flash Generation: Let you change flash generation time (milliseconds) for PBX detection.
Ring Frequency:	You can configure how long the Ring Frequency do you want to use.
DTMF tone power:	Sometimes you input DTMF, but no request. You can adjust this function, range from -6db to +6db.

FXO Transmit Hybrid:	ECHO cancellation adjust, 3 mode setting to solve ECHO problem, default is Mode 1. If you have echo problem, try to select other Option.
FXO Ringer Voltage Threshold:	Sometime you use FXO connect to PSTN/PBX can't be pick up the call. PSTN/PBX can't detection FXO voltage, you can adjust this function. Default is low.(Low : 16.5 Vrms Medium: 24 Vrms High: 49.5 Vrms)
FXO Ringer Voltage filter:	Some vendor's PBX generates the leakage voltage from extension port. That will mislead the FXO become Off-hook status. This function was set to avoiding a leakage voltage signal is detected as ring coming.
FXS Battery Reversal Generation:	Some Payphone need Battery Reversal to count call money. Gateway FXS can generation that Battery Reversal. If you don't use this function, don't enable it.
FXO Answer Supervision:	Enable battery reverse to detect polarity from PSTN line. The PSTN line can send. H.323 case: Sending the Q.931 connect signal to caller when detecting polarity reverse from PSTN Line. SIP case: Sending the 200 OK connect signal to caller when detecting polarity reverse from PSTN Line. Voice Detection: when the FXO connect PSTN/PBX, gateway will detection the PSTN/PBX send voice , it will send Q.931 connect or SIP 200 ok.
Line Silence Disconnect:	When FXO call have silence, the call will disconnect automatically. {Default is Disable}
FXO Answer Delay Time	Some PSTN/PBA when call incoming can't pick up first time. It need delay few second time. You can set FXO pick up call delay time.(default is 0)
FXO Answer Mode	FXO Answer Mode Concept: When user calls the PSTN line which was connected with the FXO port, there are three answer mode for user to configure. 4. Ringing Answer Mode (Default Setting): FXO answer the call once the ring coming from PSTN line. 5. Connecting Answer Mode: **Case A:** "Hot Line Number" was NOT assigned in the FXO port. FXO answer the call once the ring come from PSTN line. **Case B:** "Hot Line Number" was assigned and the Hot line number belongs to remote VoIP device. In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call. (Note: This case can avoid charging for the Local PSTN call when the remoting VoIP devic still ring.) **Case C:** "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway. FXO port will not answer (off-hook) till the Phone (connected to the FXS port) was picked up by user. (Note: This case can avoid the Local PSTN charge when the FXS port still ring.) 6. Non Answer Mode: FXO will NOT answer the call in any time. (Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination)

[Network Advance]



Item	Description
Smart-QoS:	If this function is enabled, when VoIP call is occurred, the other data will be automatically reduced traffic which across the internet in order to guarantee the voice bandwidth.
Bandwidth control:	You can configure your bandwidth what the Max byte of download and upload of ADSL modem rate.
G.723/G.729 Bandwidth:	Setting G.723 / G.729 voice compression size. Quality and Packet size can adjust by you want.
IP TOS:	Some Router support TOS(Type of Service), when you enable the TOS function, the router will process those packets firstly.(default is disable)

4.2.3 Direct call (Peer to Peer) setup

If you don't registered Gatekeeper or SIP proxy server, you can make call by Peer to Peer. For SIP or H.323, setting the dialing plan, and can make direct call.

Overview of the Dialing Plan

The "**Dialing plan**" need setting when the user use the method of Peer-to-Peer H.323 (SIP) VoIP call or registering H.323 Gatekeeper (SIP Proxy Server) Mode. The H.323(SIP) Dialing Plan has two kinds of directions: Outgoing (call out) and Incoming (call in).

1. **Outgoing Dial Plan:**

   Peer-to-Peer Call Mode: Effective

   Registering to H.323 Gatekeeper (SIP Proxy Server) Mode: Effective

2. **Incoming Dial Plan:**

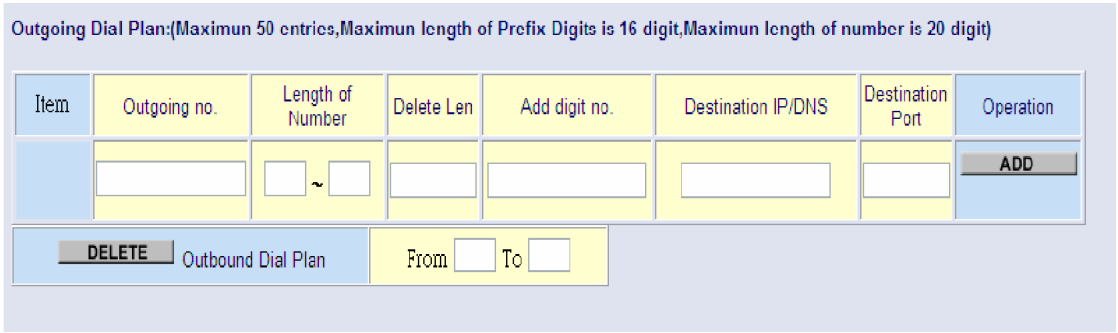
   Peer-to-Peer Call Mode: Effective

   Registering to H.323 Gatekeeper (SIP Proxy Server) Mode:

   The leading number would **register** to H.323 Gatekeeper (SIP Proxy Server)

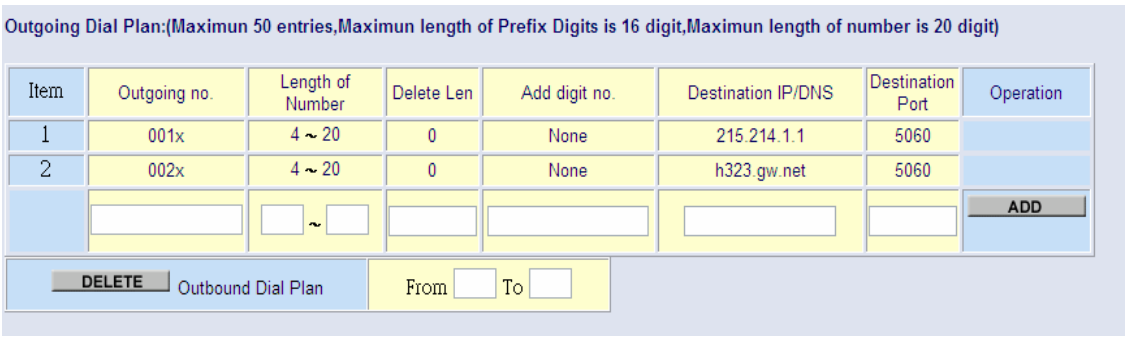
When you use direct call, you must setting your VoIP protocol firstly. Use direct call, you should setting the same protocol both of UA. Both of UA must support dial plan function. Some ATA don't support Dialing plan, it maybe let direct call failed.

In the “Outgoing Dial Plan Configurations” settings: Maximum Entries : 50



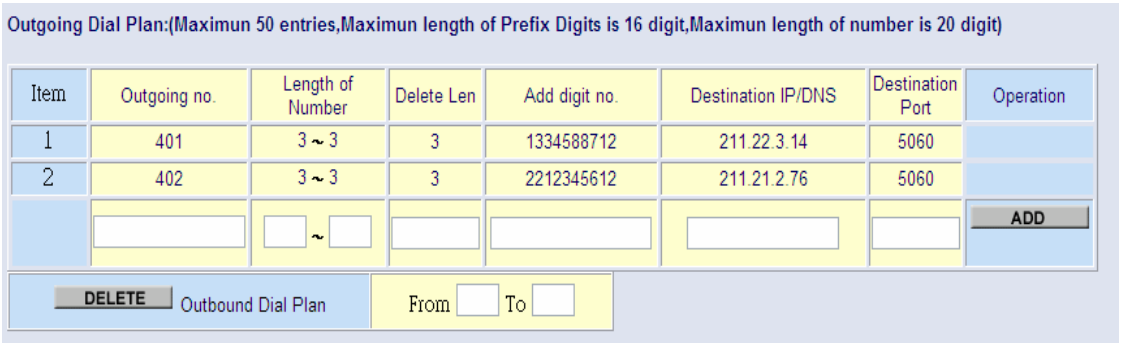
- “Outbound number” is the leading digits of the call out dialing number.
- “Length of Number” has two text fields need filled: “Min Length” and “Max Length” is the min/max allowed length you can dial.
- “Delete Length” is the number of digits that will be stripped from beginning of the dialed number.
- “Add Digit Number” is the digits that will be added to the beginning of the dialed number.
- “Destination IP Address / Domain Name” is the IP address / Domain Name of the destination Gateway that owns this phone number.
- “Destination Port” is port of the destination gateway use.(Default is 5060)

Example1: Normally Dial

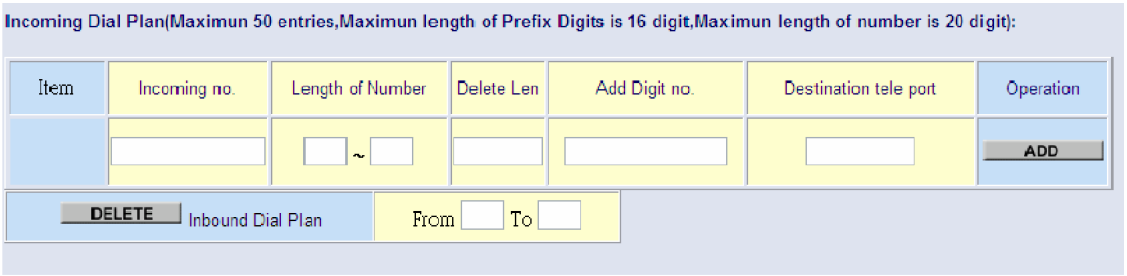


1.001x leading call out, call to Destination IP address: 211.22.3.14
2.002x leading call out, call to Destination Domain Name: h.323.gw.net

Example2: Speed Dial

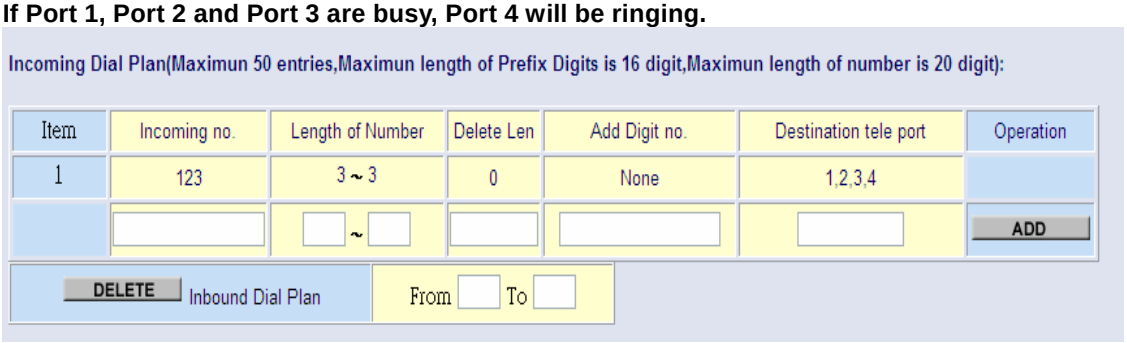


1. If user dial “401”,
Gateway automatically dial “1334588712” to Destination IP address: 211.22.3.14
2. If user dial “402”,
Gateway automatically dial “2212345612” to Destination IP address: 211.21.2.76
In the “Incoming Dial Plan Configurations” settings: Maximum Entries : 50

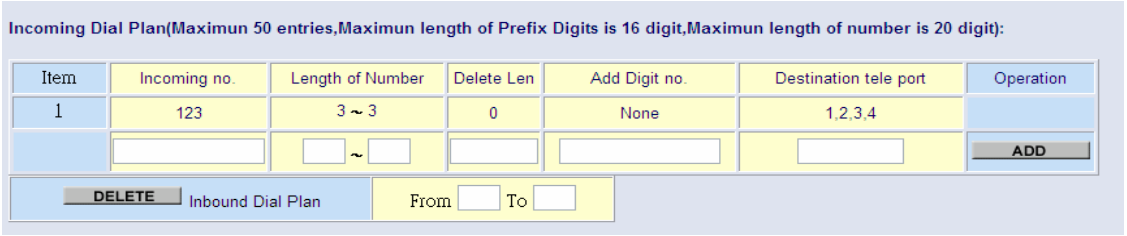


- "Inbound number" is the leading digits of the dialing number.
- "Length of Number" has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial.
- "Delete Length" is the number of digits that will be stripped from beginning of the dialed number.
- "Add Digit Number" is the digits that will be added to the beginning of the dialed number.
- "Destination Tele port" is "Tel-port"; this is for local dial plan setting phone number.

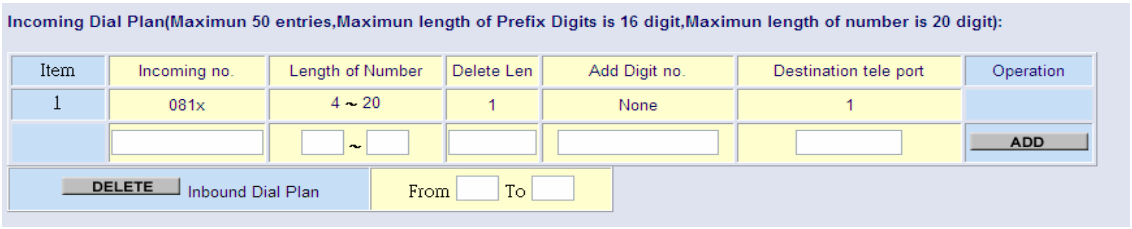
Example1: Hunting for FXS Port
Port 1: FXS
Port 2: FXS
Port 3: FXS
Port 4: FXS
H.323 (SIP) number "123"call incoming, Port 1 will be ringing.
If Port 1 is busy, Port will be ringing.
If Port 1 and 2 are busy, Port 3 will be ringing.
If Port 1, Port 2 and Port 3 are busy, Port 4 will be ringing.



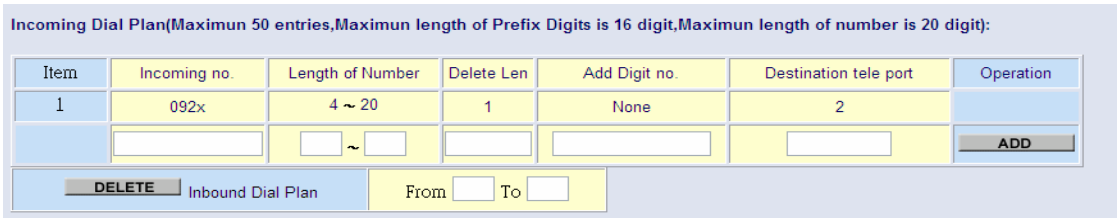
Example2: Hunting for FXO Port
Port 1: FXO was connected to PSTN.
Port 2: FXO was connected to PSTN.
Port 3: FXO was connected to PSTN.
Port 4: FXO was connected to PSTN.
H.323 (SIP) number "123"call incoming, Port 1 will be off-hook and hear the Dial Tone from PSTN.
If Port 1 is busy, Port will be will be off-hook and hear the Dial Tone from PSTN.
If Port 1 and 2 are busy, Port 3 will be off-hook and hear the Dial Tone from PSTN.
If Port 1, Port 2 and Port 3 are busy, Port 4 will be off-hook and hear the Dial Tone from PSTN.



Example3: Termination Call to FXO for One-Shoot Call
Port 1: FXO was connected to PSTN (area code is 81xxxxxxxx).
H.323 (SIP) leading number "081x"incoming, and delete the first one digit "0", and call to PSTN number.



Example4: Termination Call to FXO
Port 1: FXS
Port 1: FXO was connected to PSTN (area code is 92xxxxxxxx).
Port 1 FXS call to "092x" to PSTN, FXO port will delete the first one digit "0" and call to PSTN number.

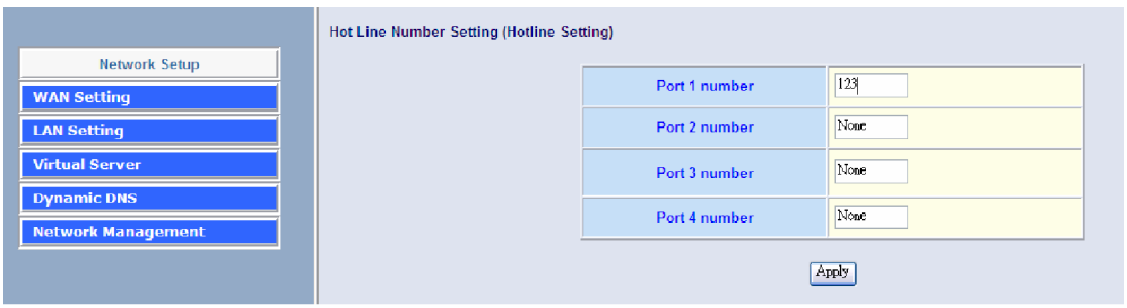


([x]: mean wild card, 0~9. Speed dial: When you dial over the number, input "#" key can speed dial call, or you key in number as the length as the "length of number")

4.2.4 Other VoIP Setting

Hot Line:

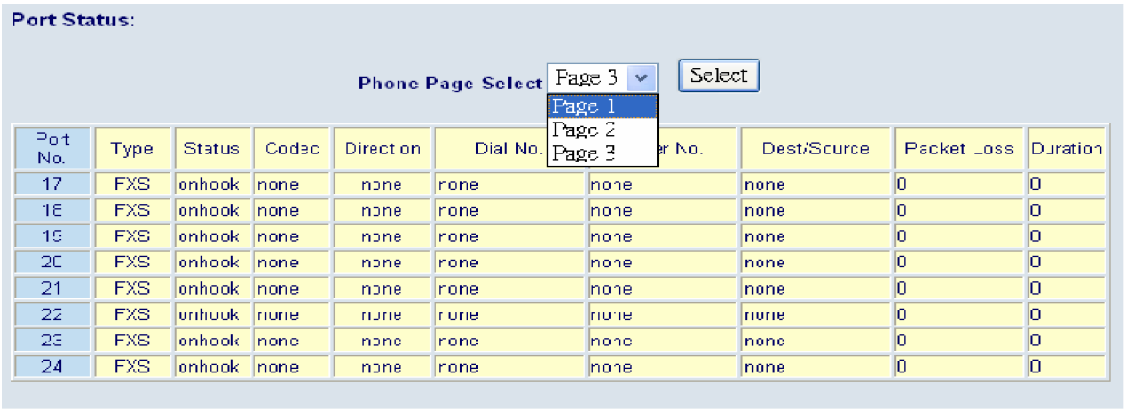
You can setting hot line. when the call incoming the hot line port, it will call hot line number automatically. The hot line call the number via VoIP, so you setting the hot line number must VoIP number. Usually, you want to FXO port call transfer to FSX, you only setting the FXO hot line to FXS number.



- **Port number:** Input FXS/FXO want to call hot line number. The call will via VoIP, so the number must be the VoIP number.

Port Status:

Each of port show status table. you can view all port status. like on/off hook , caller/callee IP, duration , and packet loss.

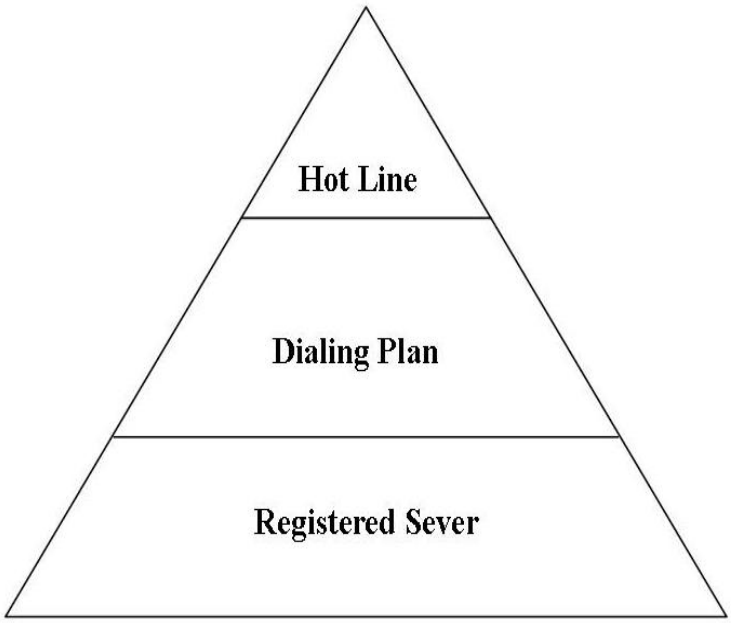

Port Status:
Phone Page Select
Page 3
Page 1
Page 2
Page 3
Select
Port No.
Type
Status
Codec
Direction
Dial No.
Dest/Source
Packet Loss
Duration
17 FXS onhook none none none none none 0 0
18 FXS onhook none none none none none 0 0
19 FXS onhook none none none none none 0 0
20 FXS onhook none none none none none 0 0
21 FXS onhook none none none none none 0 0
22 FXS onhook none none none none none 0 0
23 FXS onhook none none none none none 0 0
24 FXS onhook none none none none none 0 0

- Port Status Display : This selection will display concurrent call status of this Gateway. The status information of each voice channel includes codec, dialing number and destination IP address. The status is refreshed every 3 seconds.

Call Priority:

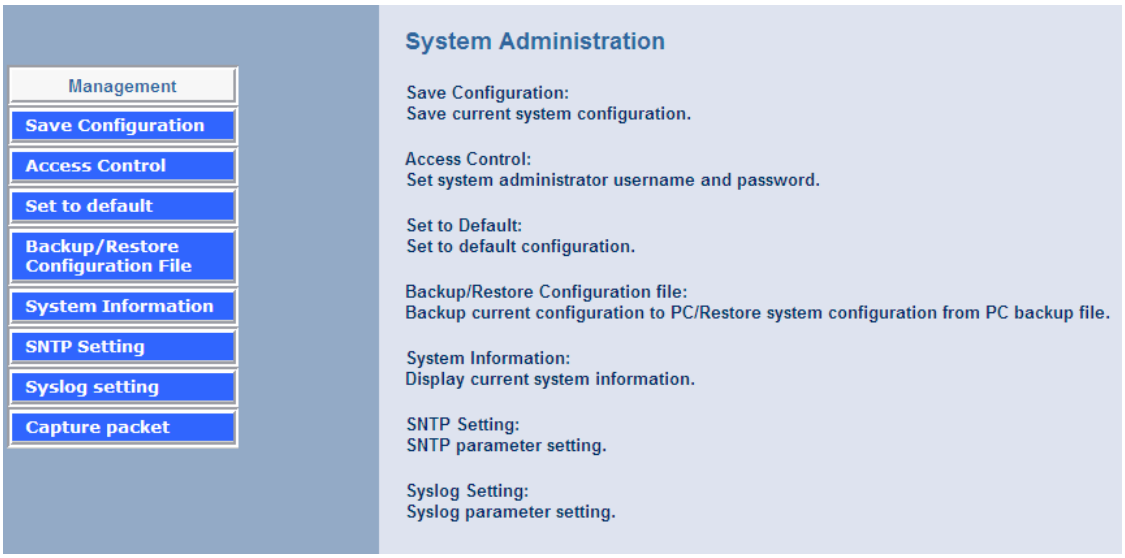
Gateway have a rule for call Priority, up to down is 1)Hot Line 2)Dialing plan 3)Registered server(SIP Proxy / H.323 Gatekeeper).When a VoIP call made, Gateway will process by Hot Line first, then it will check the dialing plan table, last fine Server(SIP/H.323).

For example, if I have a gateway , and It is registered a proxy server, I don't setting any others (Hot Line or dialing plan.). when I make a VoIP call, gateway will check Proxy server. Now,, I setting 1~2 dialing plan, and registered proxy server. When I call, gateway will check the dialing plan first, then find the proxy server. And so on…..

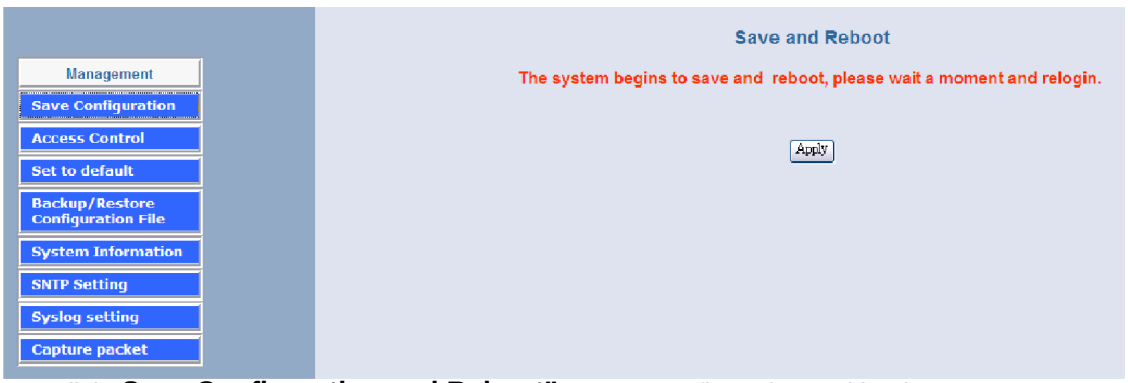


4.3 System Administrator

You can setting other gateway setting, like gateway time, Syslog that send CDR information to Syslog server, backup and restore configuration.

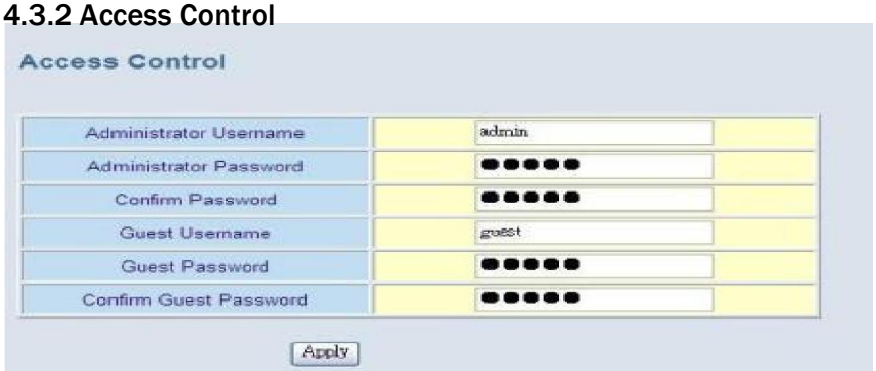


4.3.1 Save Configuration and Reboot



- Click "**Save Configuration and Reboot"** to save configuration and begin to restart. (When you set done, select "Reboot" option will auto save and reboot!)

4.3.2 Access Control



- **Changing the Administrator Password**

For security reasons, we strongly recommend that you set an administrator. password for the router. On first setup the router requires no password. If you don't set a password the router is open and can be logged into and settings changed by any user from the local network or the Internet.

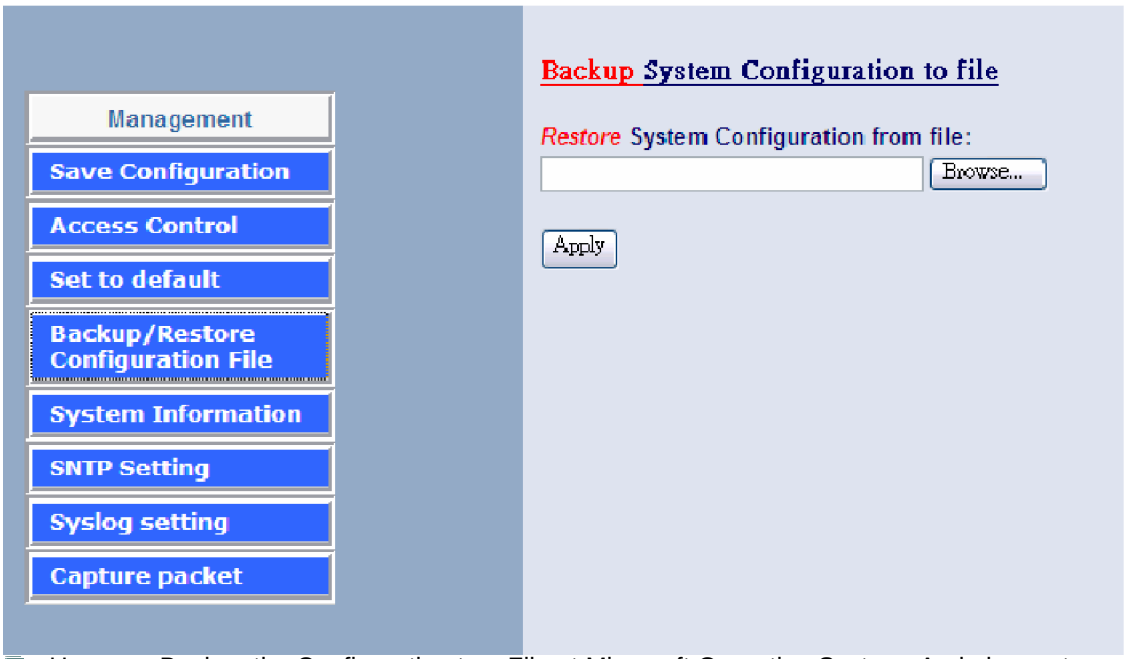
- Click **Access Control Setup**, the following screen will open.

(Guest account , if you use guest account login , you only can view gateway setting, not change and configure any gateway setting, else you login by Admin account)

4.3.3 Set To Default Configuration

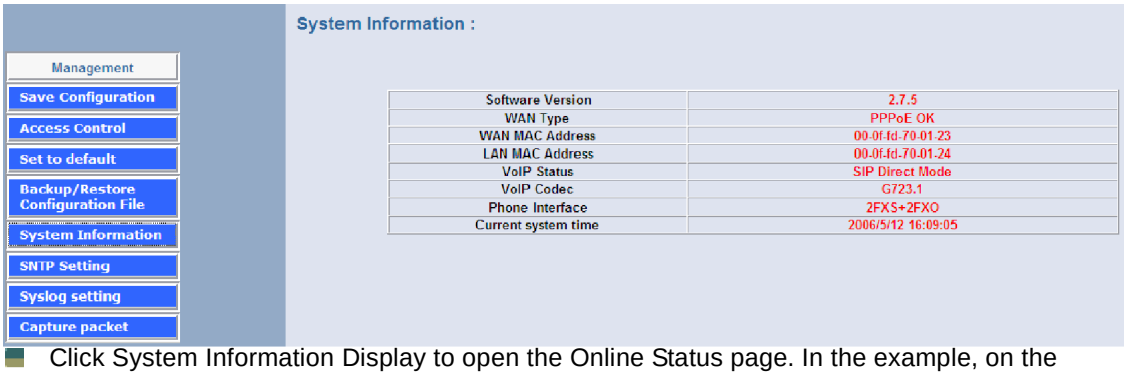
- If you want to reboot the router using **factory default configuration**, click "**Apply**" then reset the router' s settings to default values.

4.3.4 Backup/Restore Configuration to a File



- User can Backup the Configuration to a File at Microsoft Operation System. And also restore the configuration file to the GW from PC.

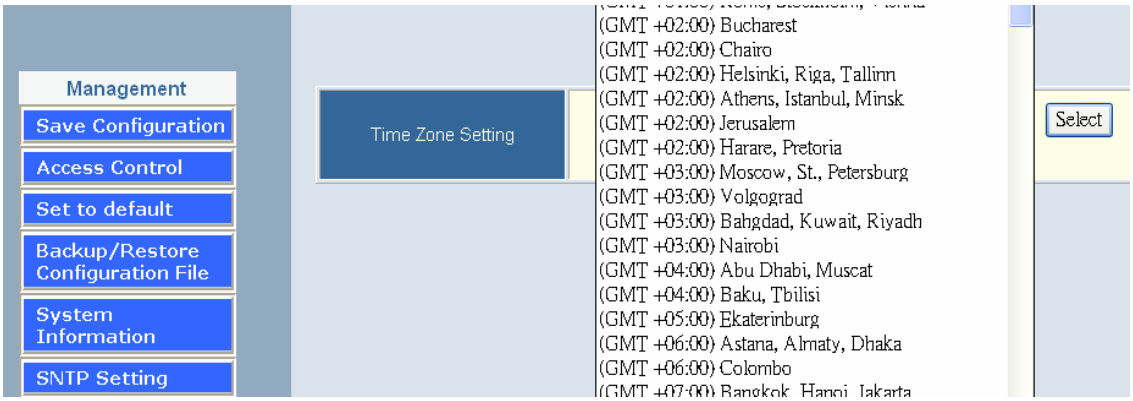
4.3.5 System Information Display Function



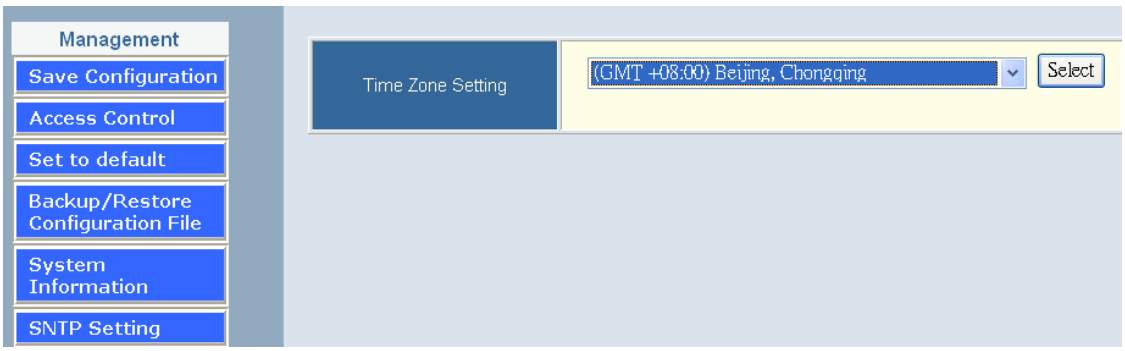
- Click System Information Display to open the Online Status page. In the example, on the following page, both PPPoE connection is up on the WAN interface, H323 Status, MAC address, Register Status, etc….

4.3.6 SNTP Setting Function

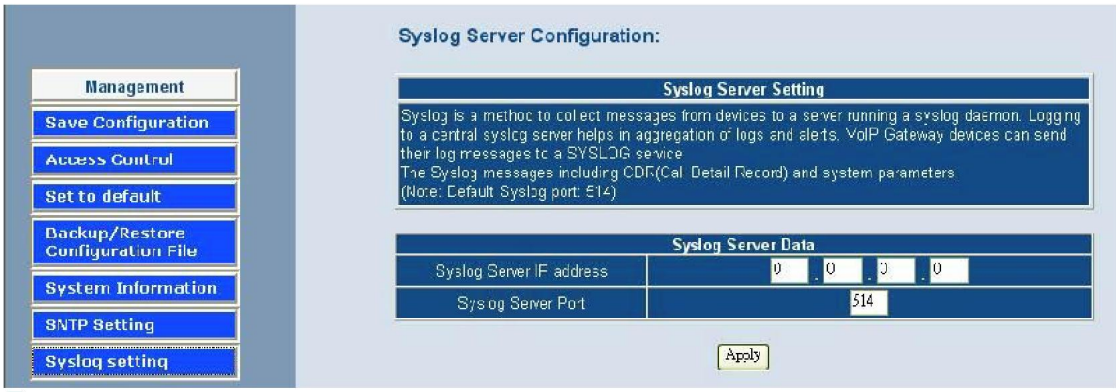
Click **SNTP Setting** to open the Online Status page. In the example, on the following page,
.



Use SNTP Setting—When checked, Gateway uses a Simple Network Time Protocol (SNTP) to set the date and time . The Gateway synchronizes the Gateway's time after you select the time zone. *Use SNTP Setting*, Select the time zone which Gateway was at.

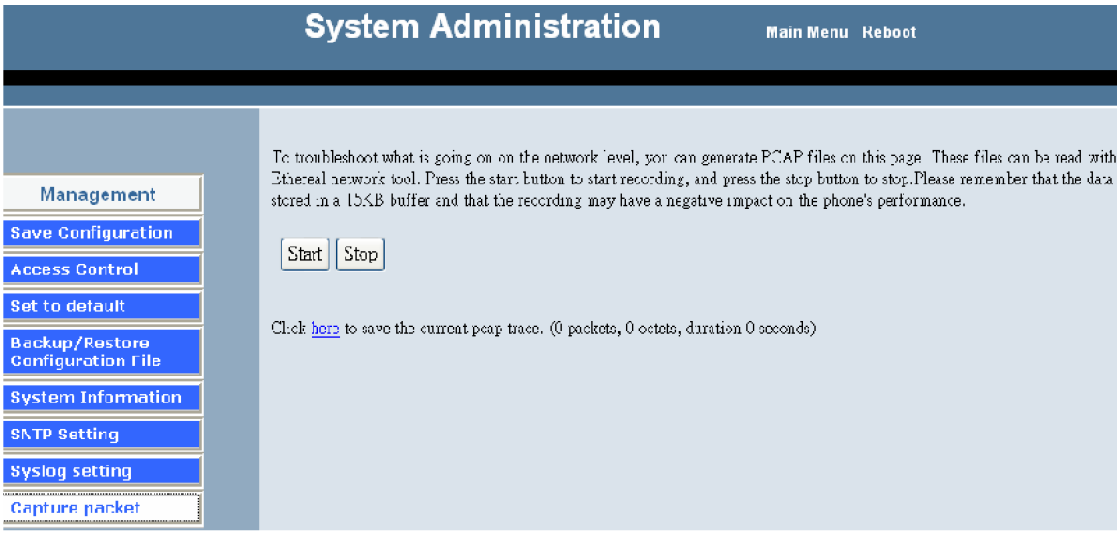


4.3.7 Syslog Setting Function


Management
Save Configuration
Access Control
Set to default
Backup/Restore Configuration File
System Information
SNTP Setting
Syslog setting
Syslog Server Configuration:
Syslog Server Setting
Syslog is a method to collect messages from devices to a server running a syslog daemon. Logging to a central syslog server helps in aggregation of logs and alerts. VoIP Gateway devices can send their log messages to a SYSLOG service.
The Syslog messages including CDR(Call Detail Record) and system parameters.
(Note: Default Syslog port: 514)
Syslog Server Data
Syslog Server IP address
0 . 0 . 0 . 0
Syslog Server Port
514
Apply

- Use Syslog server to record your Gateway log file. you can setting you syslog server IP address for this function. Syslog information include the CDR source!

4.3.8 Capture Packets Function


System Administration
Main Menu Reboot
Management
Save Configuration
Access Control
Set to default
Backup/Restore Configuration File
System Information
SNTP Setting
Syslog setting
Capture packet
To troubleshoot what is going on on the network level, you can generate PCAP files on this page. These files can be read with Ethereal network tool. Press the start button to start recording, and press the stop button to stop.Please remember that the data stored in a 150KB buffer and that the recording may have a negative impact on the phone's performance.
Start
Stop
Click here to save the current pcap trace. (0 packets, 0 octets, duration 0 seconds)

- Use “Capturer Packets” to record Gateway packets. You can start and stop the capture then save the file to PC Use the Ethereal Tool (www.ethereal.com) to analyze the packets.

(if gateway have interoperability problem, you can capture the packet, send to us . we can refer this packet to bebug.)

4.5 Update firmware

Gateway update firmware only support use **telnet** via **FTP,** no other else upgrade function.

FTP upgrade Requirement and Process

1. Environment Requirement

- PC with FTP Server (Server-U software, 3CDaemon,..)
- PC or Notebook witch connected to LAN port of Gateway.
- Put the image (firmware) named “s400.xxx” at the assigned folder in FTP Server.

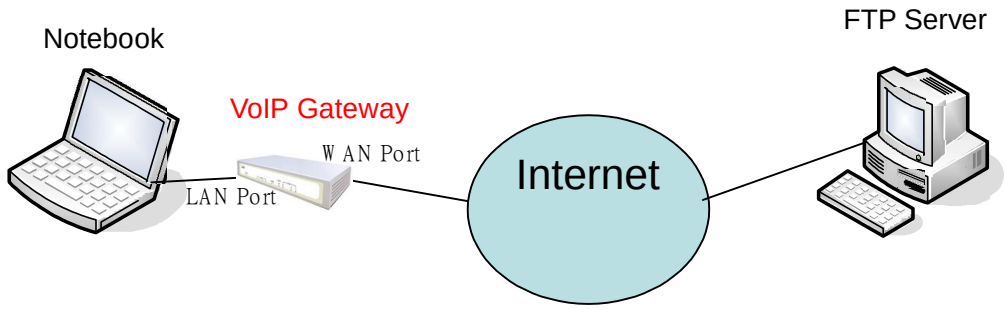
(for example: “s400.270” is version 2.7.0)

Note: Our company FTP server, you can use it to upgrade

Free FTP server : 61.218.109.83

username: share , password: 19730809

Environment Architecture (Gateway and FTP server are in Internet):



2. Upgrading Process

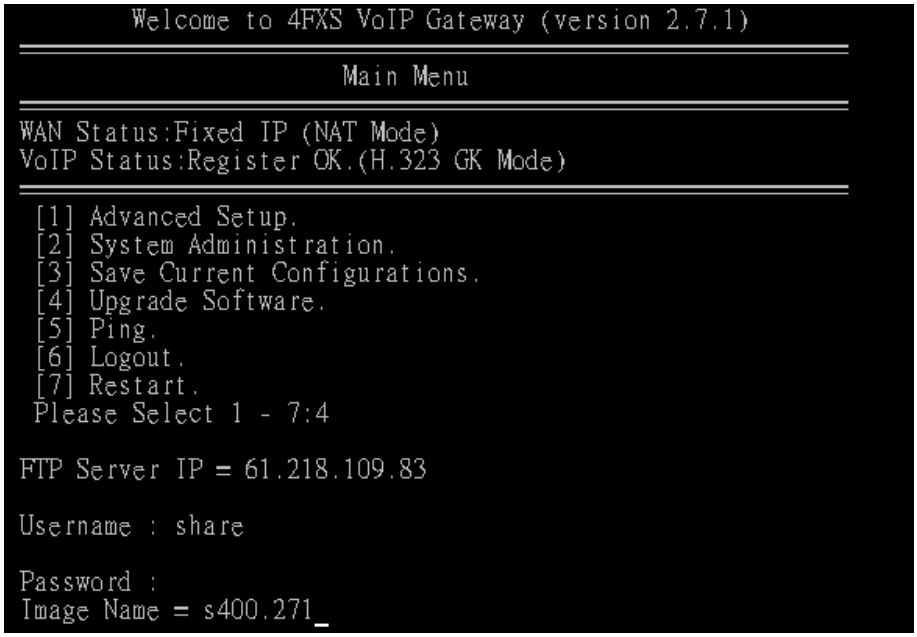
Notebook Telnet VoIP GW

[Open DOS mode]

->C:> telnet 10.10.10.1 (Default LAN port IP address) (before version 2.6.5)

->C:> telnet 222.222.222.1 (Default LAN port IP address) (after version 2.6.6)

Please select [4] Upgrade Software

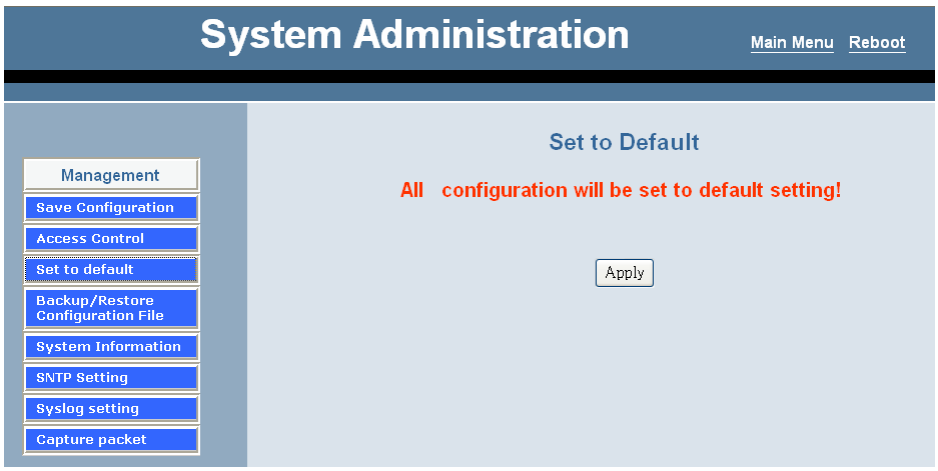


Please input IP address of FTP server like as : 61.218.109.83
Username : share
Passswd : 19730809
Imagename: s400.271
Upgrade (y/n) : y , then will write the firmware to flash.
(In different module or firmware , maybe have different change)

Connected to 61.218.109.83 port 21
[3] from 218.168.180.216 port 60002
220 (vsFTPd 1.2.0)
[Command] USER share
331 Please specify the password.
[Command] PASS xxxxxx
230 Login successful.
receiving bvw.15
[Command] TYPE I
200 Switching to Binary mode.
[4] going to listen 218.168.180.216 port 60002
[Command] PORT 218,168,180,216,234,99
200 PORT command successful. Consider using PASV.
[4] listener 0.0.0.0 port 60003
[Command] RETR bvw.15
150 Opening BINARY mode data connection for bvw.15 (1173940 bytes).
[4] Socket closed.
[5] accept from 61.218.109.83 port 20
Starting the file transfer
1173940 bytes received in 39915 ms, (29.41Kbytes/sec), transfer succeeded
[5] Socket closed.
226 File send OK.
[3] Socket closed.
Upgrade(y/n) : y

After writing flash, Please reboot the Gateway.
If the new firmware (image) was most different with the previous version, please **push** the

hardware reset bottom to set to default.
If the VoIP Gateway is in remote site, please use WEB configuration to **set to default.**



Solwise Ltd
E-mail: sales@solwise.co.uk
Web Site: http://www.solwise.co. uk

Appendix

A FAQ List

1. What is the default administrator password to login to the gateway?
A: By default, your default username is “admin”, default password is “admin” to login to the router. For security, you should modify the password to protect your gateway against hacker attacks.

2. I forgot the administrator password. What should I do?
A: Press the **Reset** button on the rear panel for over 5 seconds to reset all settings to default values. Default username / password is admin / admin.

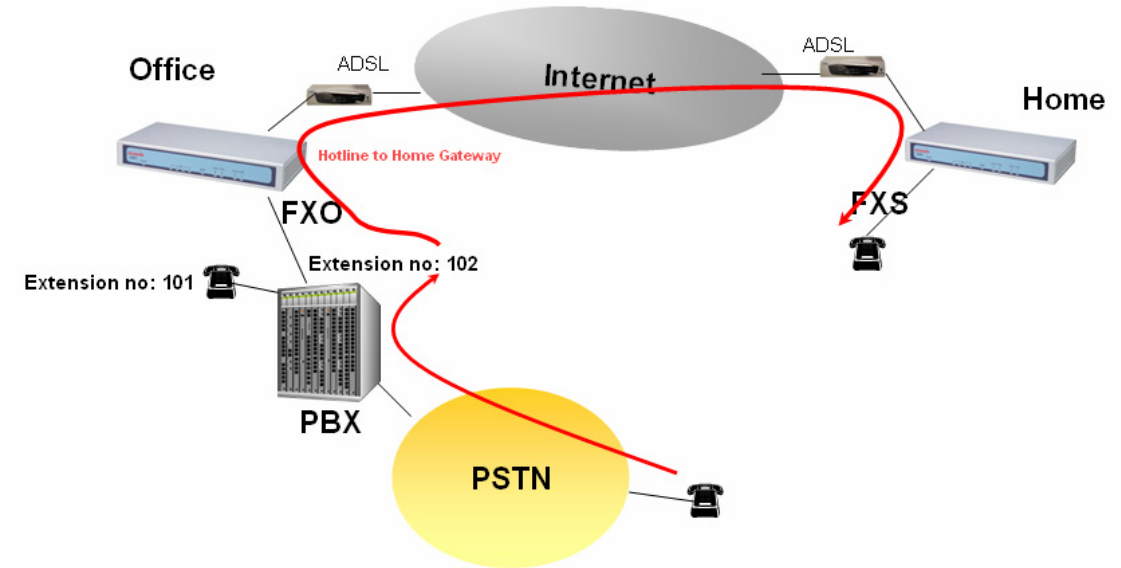
3. What is the default IP address of the router?
A: The default WAN IP address is 192.168.1.1 with subnet mask 255.255.255.0.
The default LAN IP address is 222.222.222.1 with subnet mask 255.255.255.0.

4. Why is it that I can ping to outside hosts, but not access Internet Web sites?
A: Check the DNS server settings on your PC. You should get the DNS servers settings from your ISP. If your PC is running a DHCP client, remove any DNS IP address setting. As the router will assign the DNS settings to the DHCP-client-enabled PC.

5. What is the maximum number of IP addresses that the DHCP server of the gateway can assign to local PCs?
A: The built-in DHCP server can support 253 IP addresses for local network usage.

6. What is different [set to default] and [Factory set to default]?
A: Factory set to default, you must push RST button until 5 second, gateway will clear all your setting, and let gateway Wan/Lan port become the factory default(192.168.1.1/222.222.222.1). When you use setting to default by Web or telnet , it will clear all your setting, but the wan port setting will be saved. If you remote the gateway, after set to default, you can login gateway again. No reset the gateway wan port again.

7. What is the Flash?



1. PSTN Call from PSTN to Office PBX and dial the extension 102 go to Gateway
2. Call to Gateway of Home by Hotline.
3. Home user needs call transfer to extension number 101.

4. Dial Flash and Gateway FXS detect and generate the Flash to PBX in Office.
Flash is mean on-hook and off-hook fast switch, on-hook and off-hook duration. It usually use on PBX system, in place of transfer function key.

8. Why can I call out when the gateway under the NAT?
A: VoIP product almost have NAT Pass through problem. By SIP, there are many NAT Pass through Function can solve 80% NAT Problem. You can choose STUN/Outbound Proxy/ Symmetric RTP to Pass through NAT, you don't set any other setting (DMZ/Virtual Server) by router side. If you use STUN/Outbound Proxy, you must have a STUN/Outbound Proxy Server to support. If they can't pass NAT, please open the DMZ/Virtual Server by Router/NAT/Firewall.

9. Why does the one way talk happen?
A: Generally, one way talk happen when use the different codec between VoIP device make call. Please check and setting the same codec, most one way talk will be solved.

10. Why can I call out by Gateway?
A: Please chick your Gateway is registered SIP Proxy Server (ITSP), and chink your Internet works fine. Gateway can't make a call without Internet or SIP Account that from ITSP supply. You must have a SIP account or know the other Gateway IP/Domain Name, then you can make a VoIP call.

11. I can't use web Interface to setting gateway. How can I do?
A: Please check you PC connect the Gateway Lan port or PC and ATA with the same Subnet. If you PC aren't at the same Subnet, you can't Login the gateway Web interface. Else you let your gateway on Public Internet(Public IP address).when you Lan and Wan can't login always, try to set factory default. (Push RST button until 5 second), and try again

12. Why I use asterisk by G.729 sometimes disconnect happen?
A: In asterisk setting VAD must disable, if you open Silence Compression (VAD), it will make call disconnect happen, please disable the option when you use the asterisk.

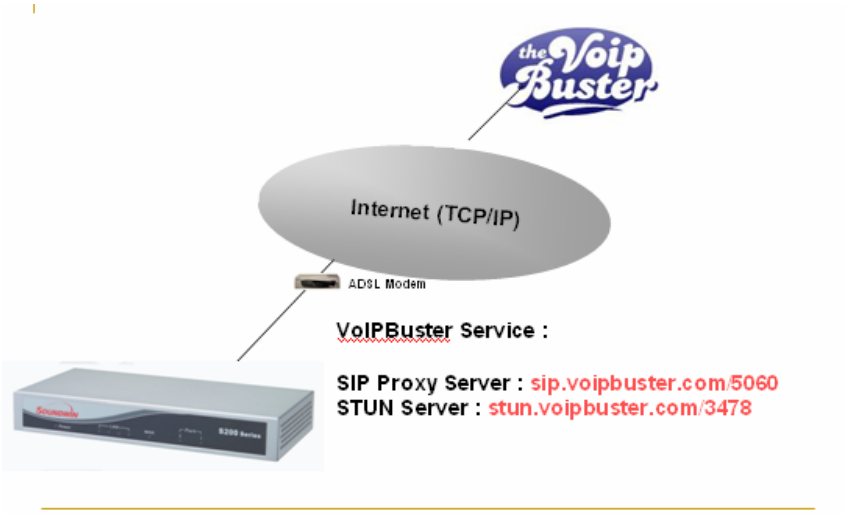
13. Can I use re-invite function when server is asterisk?
A: No, our gateway not support this function complete, the new firmware will support re-invite.

14 Why can i register and use after setting?
A: After setting, please save configuration and reboot, after reboot you can use new configuration.

15. Why I can't use FAX?
A: The VoIP FAX has 2 mode, one is T.30, another is T.38. T.30 use voice date to FAX, so usually T.30 FAX by VoIP is work. There are some problem with T.38. Because T.38 is use protocol and data packet to sand FAX, different platform and device maybe have different change, that made the T.38 filed. If you can use T.38 to FAX, Please try to use T.30.

B SIP Setting VoIPBuster

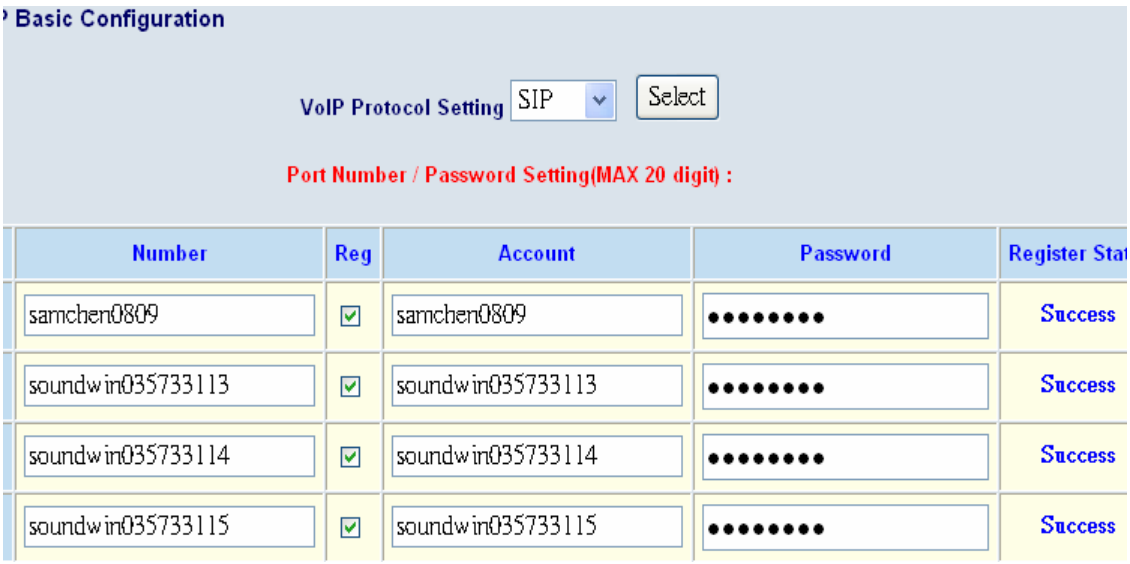
VoIPBuster Service Using VoIP Gateway



The L201 VoIP Gateway can register to VoIPBuster (http://www.voipbuster.com) VoIP service by SIP protocol and also can call SIP calls by VoIPbuster (http://www.voipbuster.com) service.

- **Gateway Setting**

1. **VoIPBuster SIP Proxy Server :** sip.voipbuster.com / 5060
2. **VoIPBuster STUN Server:** stun.voipbuster.com / 5060
3. VoIP Basic -> Setting SIP accounts and Set the Proxy Server and STUN server.



SIP Proxy Setting :

Domain/Realm	sip.voipbuster.com
SIP Proxy Server	sip.voipbuster.com/5060 ☐ use net2phone
Register Interval (seconds)	900
SIP Authentication	◉ Enable ○ Disable
Outbound Proxy Server	0.0.0.0/0

NAT Pass Setting:

NAT Pass Method	◉ STUN ○ Symmetric RTP
STUN Server address	stun.voipbuster.com
STUN Server port	3478

- **How to dial the call?**

  00 - country code – area code

  For example soundwin company phone number is +886-35733113, the dial number is 0088635733113

- **VoIPBuster Provides Free Land Line (Fixed Line) Calls**

FREE CALLS! (LANDLINES ONLY)

Andorra	Georgia	New Zealand
Australia	Greece	Norway
Austria	Hong Kong	Panama
Belgium	Iceland	Peru
Bulgaria	Ireland	Portugal
Canada	Italy	Puerto Rico
Chile	Japan	Singapore
Colombia	Latvia	Slovenia
Croatia	Liechtenstein	South Korea
Cyprus	Luxembourg	Spain
Denmark	Malaysia	Taiwan
Estonia	Monaco	Thailand
Finland	Mongolia	Venezuela
France	Netherlands	

100% Free, no call setup! Click here for more info. For all other rates, click here

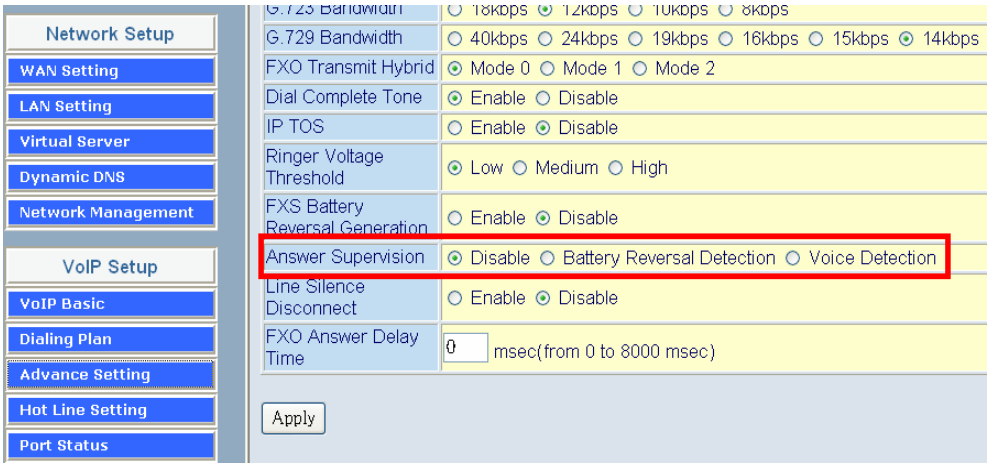
C Answer supervision

This is designed to help explain and resolve issues of answer supervision from a switch or PSTN provider that could result in billing for termination calls.

Gateway provides 2 Types of Answer Supervision:

1. Loop-Start Reverse Battery, Reverse battery (also called Polarity Reverse) is when the PSTN provider reverses the polarity of the battery voltage, for both answer supervision and disconnect supervision.

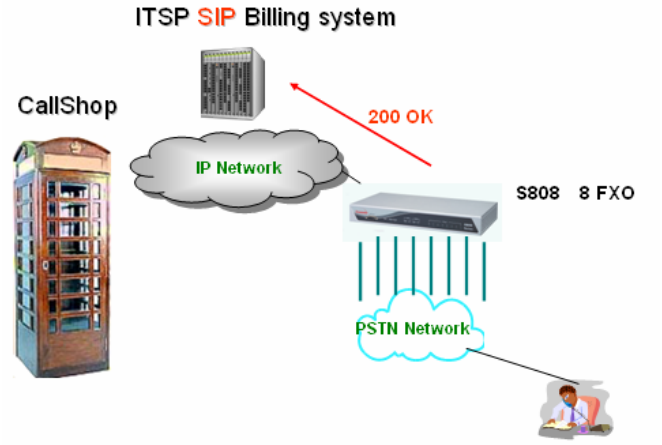
2. Voice Detection-based answer supervision is a feature where the Gateway can be configured to “listen” on the line for different tones and voice. The Gateway sends a “connect” signals out or “disconnect” signaling using internet.



SIP Application

Case 1: Loop Start Reverse Battery : PSTN Line was set Polarity Reverse:

The gateway can send the “200 OK” SIP signals to Billing System of ITSP, after the user pick up the Phone and detect the PSTN line answer voltage.



The gateway can send the “200 BYE” SIP signals to Billing System of ITSP, after the user hang up the Phone and detect PSTN line disconnect voltage.

Case 2: Voice Detection based on answer supervision:

PSTN Line was not support Polarity Reverse: The gateway can send the 200 OK SIP signals to Billing System of ITSP, after the user pick up the Phone and detect the voice.

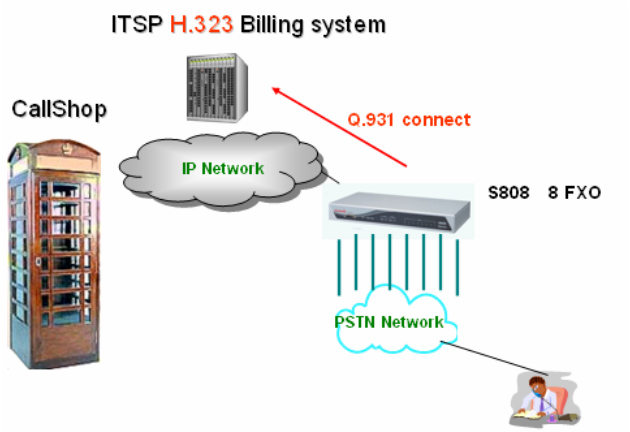
The gateway can send the 200 BYE SIP signals to Billing System of ITSP, after the user hang up the Phone and detect the hang up voice.

This type of answer supervision is not 100% accurate. Any voice frequency is detected as connect, including any intercept or recorded messages.

H.323 Application

Case 1: Loop Start Reverse Battery PSTN Line was set Polarity Reverse:

The gateway can send the “Q.931 connect” H.323 signals to Billing System of ITSP, after the user pick up the Phone and detect the PSTN line answer voltage.



The gateway can send the “Q.931 Release” H.323 signals to Billing System of ITSP, after the user hang up the Phone and detect PSTN line disconnect voltage.

Case 2: Voice Detection based on answer supervision

PSTN Line was not support Polarity Reverse:

The gateway can send the Q.931 connect H.323 signals to Billing System of ITSP, after the user pick up the Phone and detect the voice.

The gateway can send the Q.931 Release H.323 signals to Billing System of ITSP, after the user hang up the Phone and detect the hang up voice.

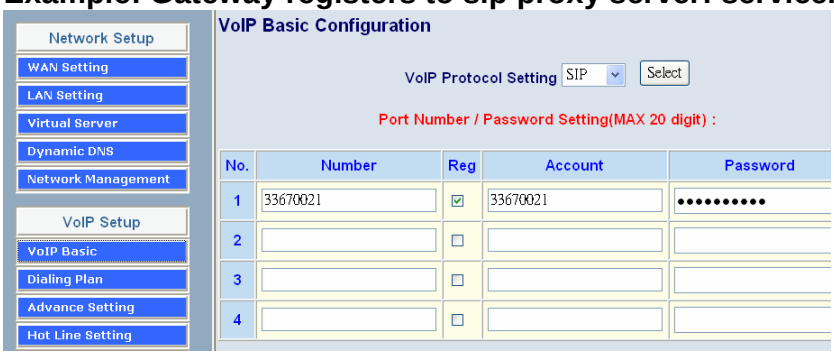
This type of answer supervision is not 100% accurate. Any voice frequency is detected as connect, including any intercept or recorded messages.

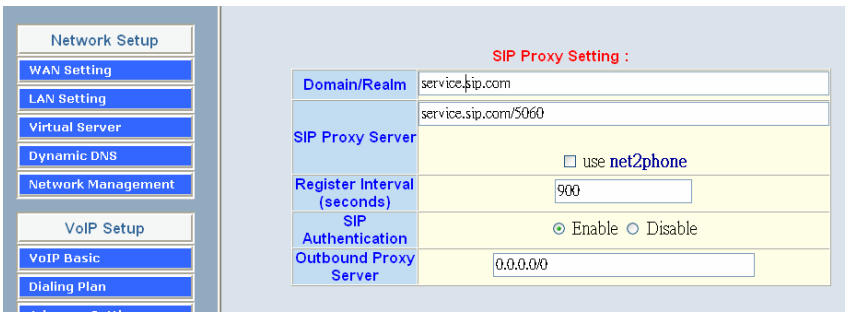
D Sip Speeds call

Speed Call Concept:
Cut your phone number down to fewer digit dialing!
Life is moving fast - you've got to dial fast. Now you can with Speed Dial. Dial the people you call most with just dialing fewer digits instead of dialing the full phone number.

- SIP RegisterMode

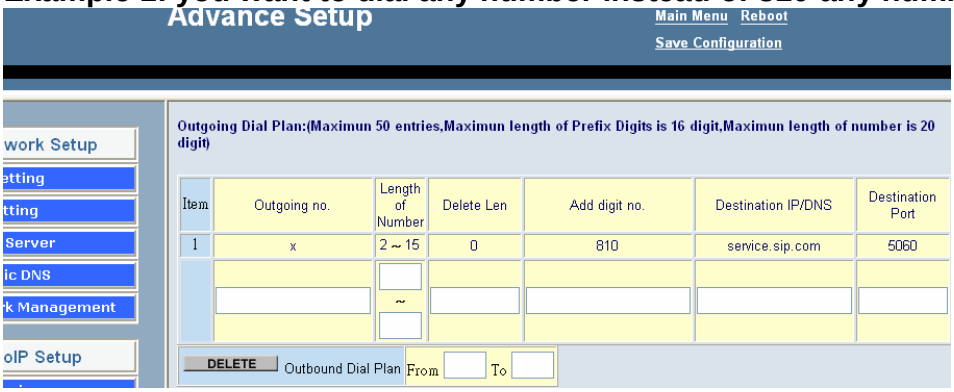
Example: Gateway registers to sip proxy server: service.sip.com





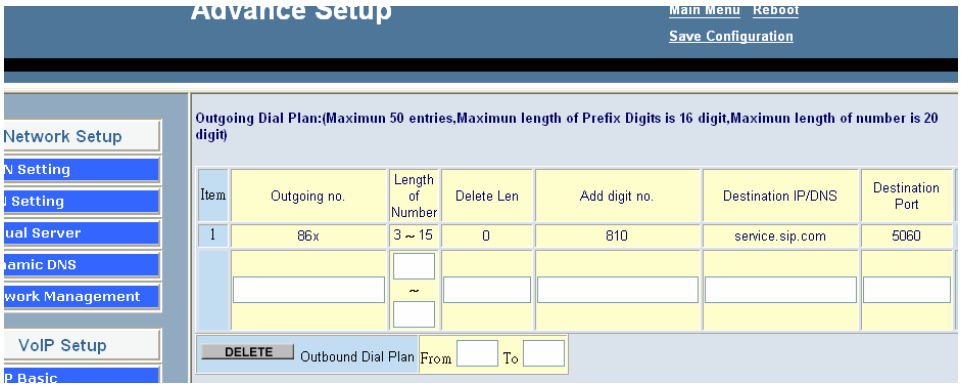
What's even better is that you can customize and manage your speed dial phone numbers in Dial Plan Setting on your gateway! Dial Plan allows you to set up to speed dial numbers that can be called with the fewer numbers.

Example 1: you want to dial any number instead of 810-any number



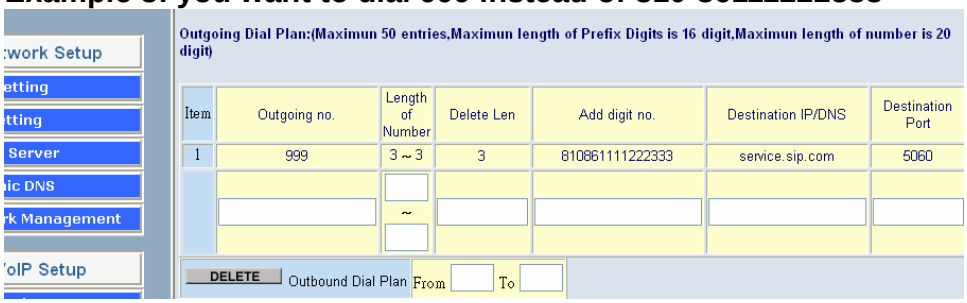
The destination IP address is the domain name of sip proxy server

Example 2: you want to dial 86-1111222333 instead of 810-86-1111222333



The destination IP address is the domain name of sip proxy server

Example 3: you want to dial 999 instead of 810-86111222333



The destination IP address is the domain name of sip proxy server

E Interoperability List

- **Gatekeeper**
  - **GnuGK openH323**
  - **Radvision ProLab GateKeeper Simulator, Version 1.0, October 2001**
  - **Clarent Gatekeeper**
  - **MediaDigm-SureKeepe**
  - **Lucent -i Merge GK**

- **SIP Proxy Server**
  - **Vovida SIP Proxy Server**
  - **SER (SIP Express Router)**
  - **Party SIP ServerV0.5.0**
  - **Clarent SIP server**
  - **Asterisk 0.5.0**
  - **SXi SIP server**

- **Gateway:**
  - **Welltek 2~8 port GW**
  - **ACCEL 2~8 port GW**
  - **Cisco 5300 and Cisco 5350 Trunk GW**
  - **Cisco ATA -186**
  - **MOSA 4 port**
  - **CISCO ATA-186**
  - **Quintum Tenor A400/A800, AS/AX Series**
  - **Antek 2500 series and vsp5004 series**
  - **Audiocodes MP-104 FXS**
  - **Clarent CPG-101 , CPG-22102Sc**
  - **D-Link 4-port gateway**
  - **BOSaNOVA Analog Gateways**

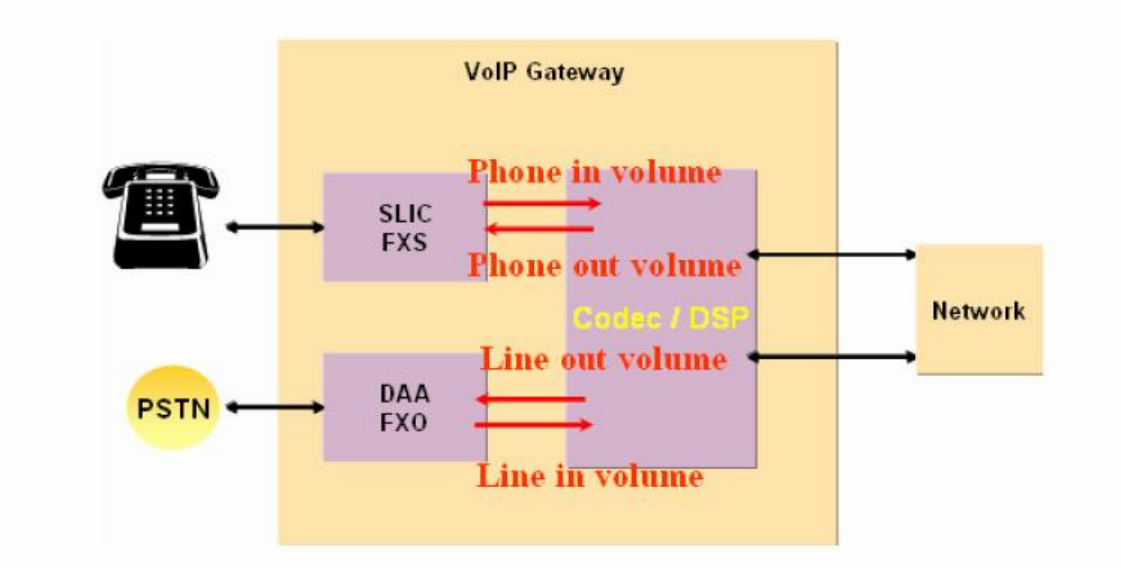
- **Trunk Gateway:**
  - **Cisco AS5300, 5350**
  - **Clarent BHG2500**
  - **AudioCodes IPM-260 Board**

- **IP Phone:**
  - **ACT LAN phone**
  - **TECOM IP PHONE**
  - **BCM IP Phone**
  - **Cisco 7900 IP Phone**
  - **UMEC video phone**

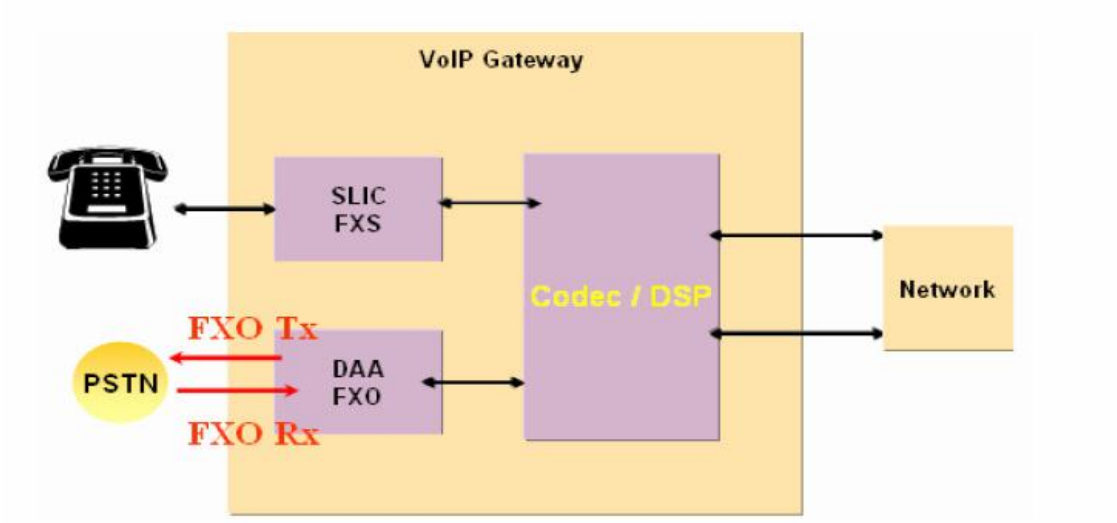
F Gateway value Setting

Phone Volume / Line Volume

Supports user can set DSP volume.



FXO TX /RX Gain



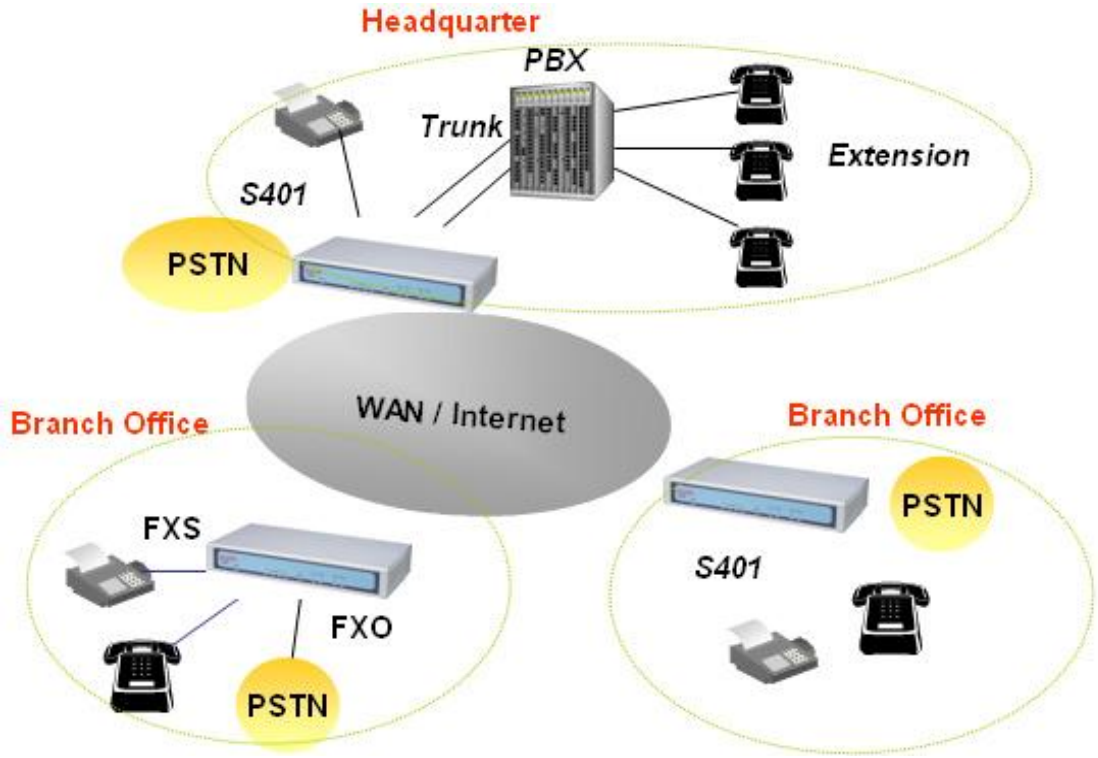
Supports multiple levels of gain and attenuation for the transmit and receive paths of DAA.

1. The FXO tx gain enable gain or attenuation in 1 dB increments for the transmit from DAA site outgoing PSTN site.

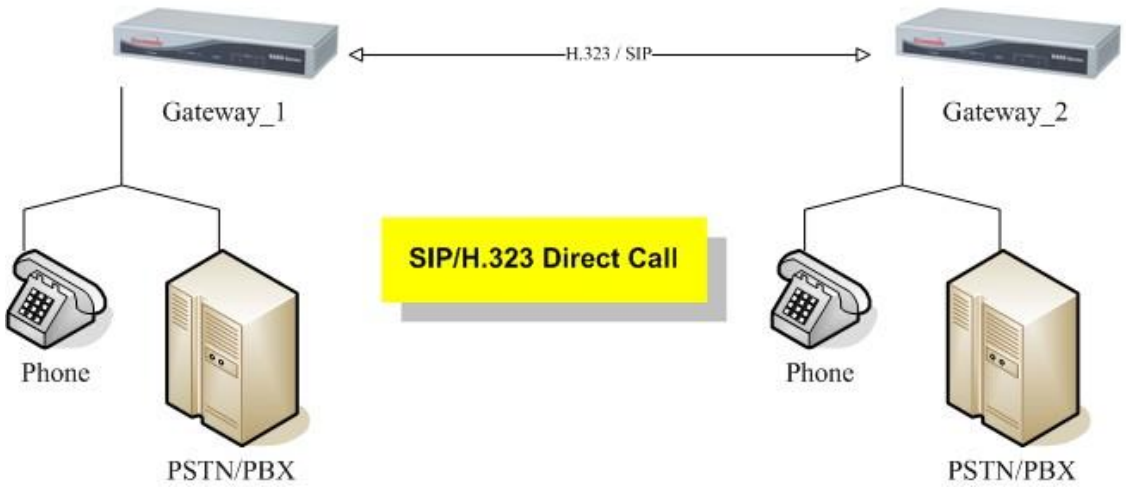
2. The FXO rx gain enable gain or attenuation in 1 dB increments for the receive from PSTN site incoming DAA site.
Note: DAA : Direct Access Arrangement.

G Scenario Application Samples

Scenario 1:
The following application was use in Multi-Office as an example



Scenario 2:
SIP or H.323 Direct call Setting



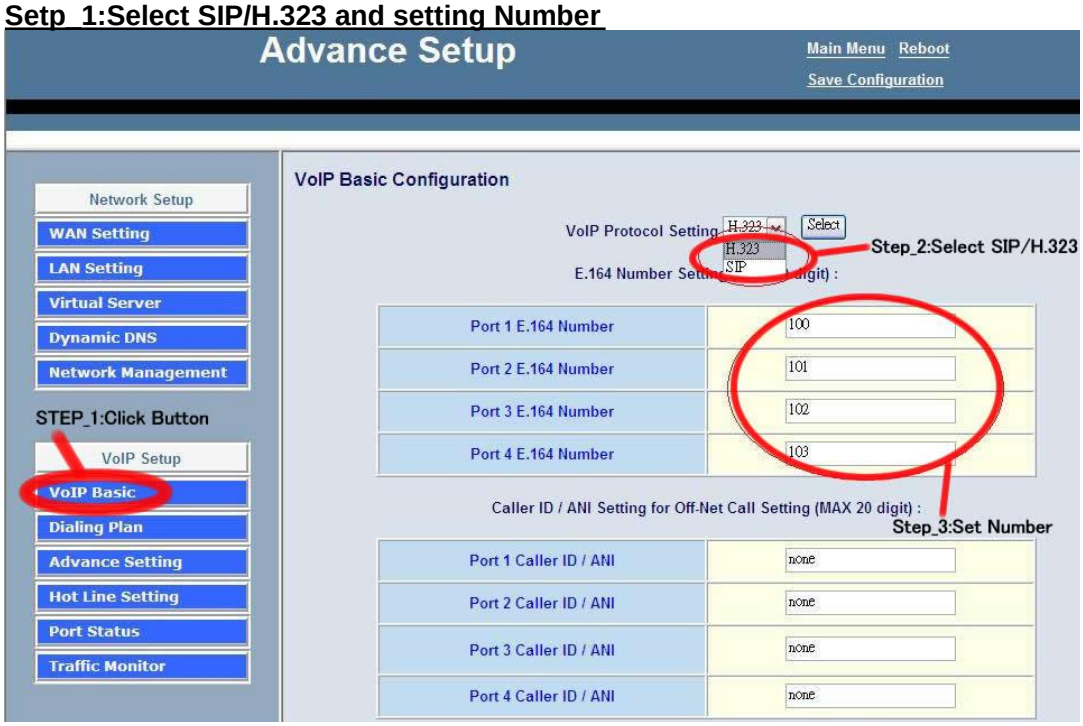
H.323/SIP How to Setting Peer to Peer mode:
-Environment Setting for

Demo- Two Gateways
Gateway Setting

Information\device	Gateway_1	Gateway_2
IP Address	IP:192.168.1.1	IP:192168.1.2
Port Number	Port_1~Port_4:100~103 (Phone Number)	Port_1~Port_4:200~203 (Phone Number)

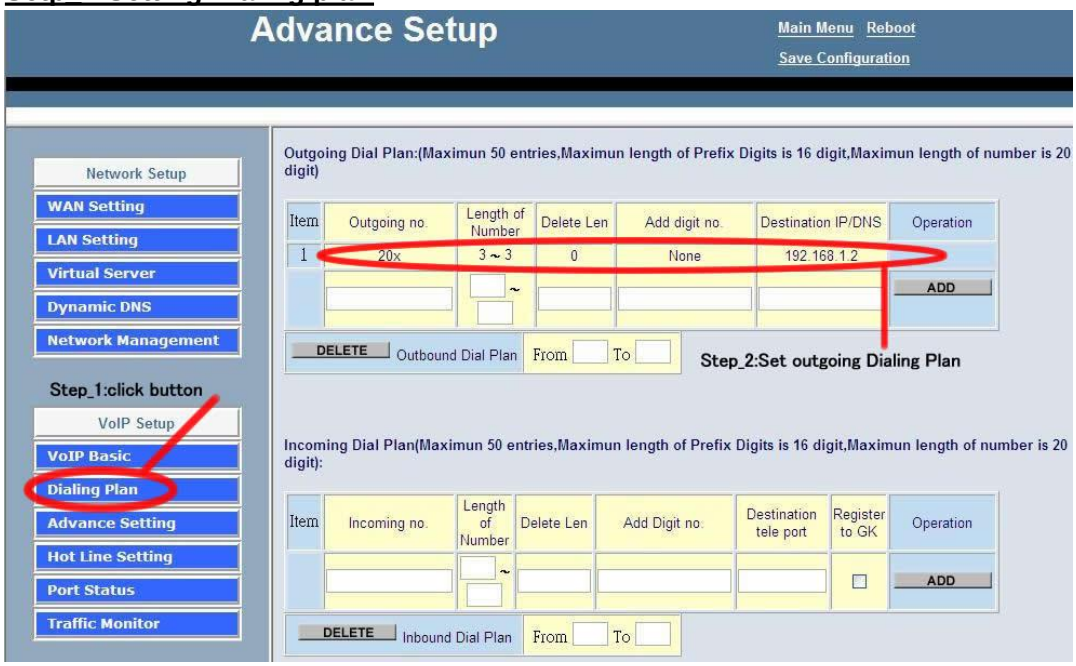
Attentions: If you want to use P2P (Direct mode), please select the same protocol SIP/H.323 both of the gateway. If you want to use different band gateway, please check the other band gateway have support Dialing Plan, we don't premise all of other band gateway can complete support P2P mode.,

Setp_1:Select SIP/H.323 and setting Number



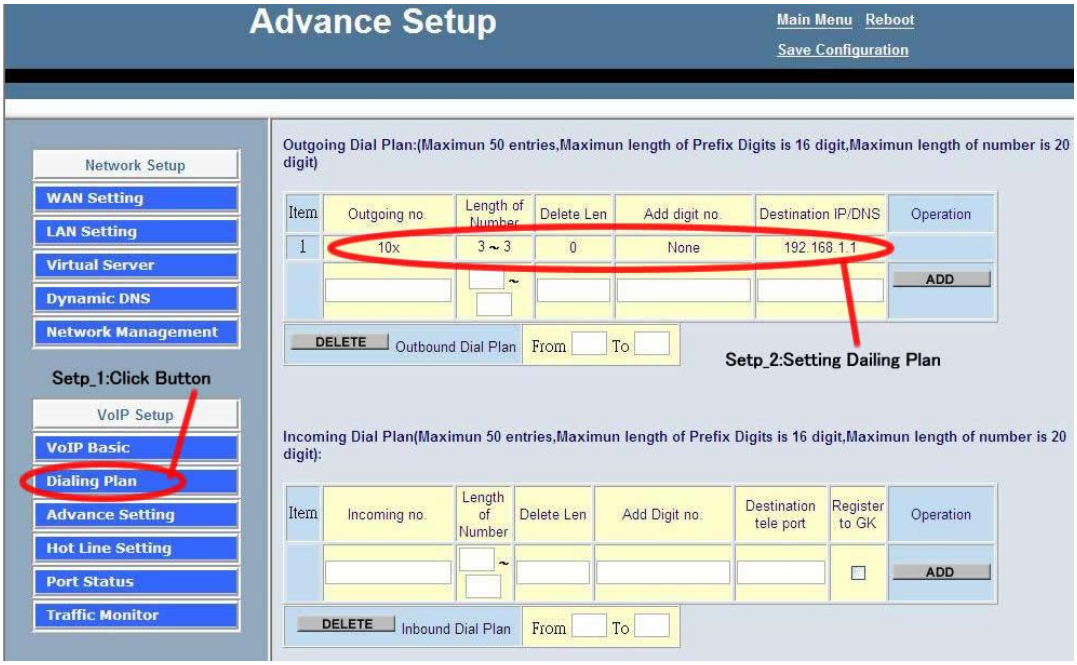
1. **Choose "VoIP Basic". Login in web interface, and in "Advance Setting".**
2. **Select you wan to use protocol (SIP/H.323).**
3. **Input you want to use call number.**

Setp_2:Setting Dialing plan



For Gateway_1 Setting

1. **Choose “Dialing plan” and Setting Outgoing Dial plan.**
2. **Setting dial plan just like picture for demo.”20x” the “x” mean wild card , it can be one of “0~9” number. And length “3~3”, when you input 3 number and the call will be made. Destination is the Gateway_2 IP address.**



For Gateway_2 Setting

1. **choose “Dialing plan” and Setting Outgoing Dial plan.**
2. **Setting dial plan just like picture for demo.”10x” the “x” mean wild card , it can be one of “0~9” number. And length “3~3”, when you input 3 number and the call will be made. Destination is the Gateway_1 IP address.**

Step_3:make call each other

1.When you setting 2 gateway done, you can make call by each other. On gateway_1, just call “200”,and the gateway_2 Port_1 will ringing, then be made a call. And gateway_2 call “100” , the gateway_1 will ringing, then be made a call.

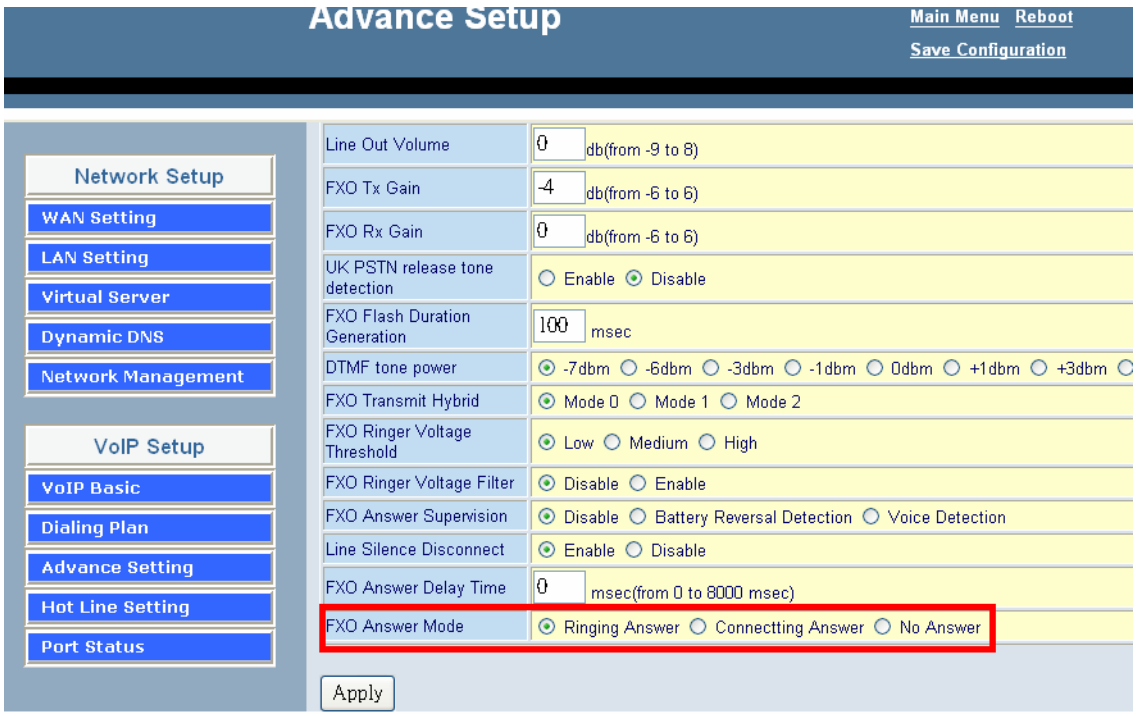
H FXO Answer Mode

FXO AnswerMode Setting

FXO Answer Mode Concept:
When user calls the PSTN line which was connected with the FXO port, there are three answer mode for user to configure.

7. Ringing Answer Mode (Default Setting): FXO answer the call once the ring coming from PSTN line.
8. Connecting Answer Mode:
   Case A: "Hot Line Number" was NOT assigned in the FXO port. FXO answer the call once the ring come from PSTN line.
   Case B: "Hot Line Number" was assigned and the Hot line number belongs to remote VoIP device.
   In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call. (Note: This case can avoid charging for the Local PSTN call when the remoting VoIP devic still ring.)
   Case C: "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway. FXO port will not answer (off-hook) till the Phone (connected to the FXS port) was picked up by user.
   (Note: This case can avoid the Local PSTN charge when the FXS port still ring.)

9. Non Answer Mode: FXO will NOT answer the call in any time.
   (Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination)

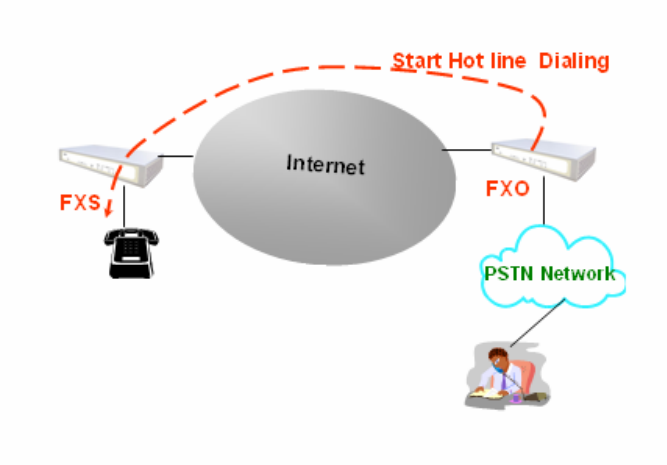


SIP Call Connecting AnswerMode

Case B: Hot Line Number" was assigned and the Hot line number belongs to SIP device.

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to remote SIP gateway
2. The phone of remote SIP gateway start ring.

3. When the phone was picked up, the remote SIP Gateway sends “SIP 200 OK” signal to FXO port.
4. Once FXO port receive the “SIP 200 OK” signal, FXO port would off-hook to answer the PSTN call.



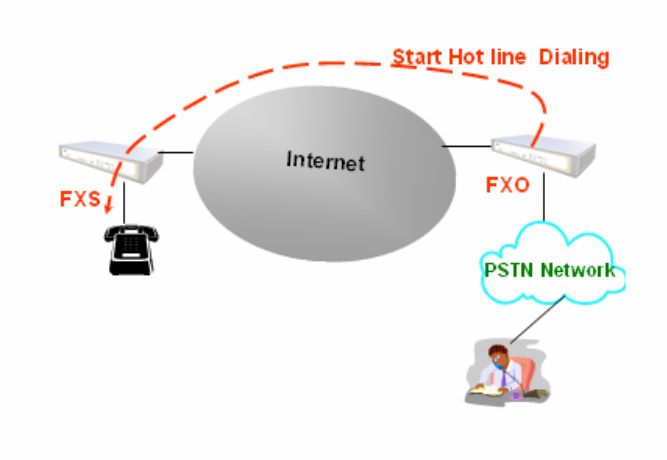
Case C: “Hot Line Number” was setting and the Hot line number was assigned to another FXS port in same Gateway.

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to FXS port.
2. The phone start ring.
3. Once the phone was picked up, FXO port would off-hook to answer the PSTN call.

- H.323 Call Connecting Answer Mode

Case B: Hot Line Number” was assigned and the Hot line number belongs to remote H.323 device.(**Note: The remote H.323 device need Disable the “Auto Answer”**)

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to remote H.323 gateway
2. The phone of remote H.323 gateway start ring.
3. When the phone was picked up, the remote H.323 Gateway send “Q.931 connect” signal to FXO port.
4. Once FXO port receive the “Q.931 connect” signal, FXO port would off-hook to answer the PSTN call.



Case C: "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway.

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to FXS port.
2. The phone start ring.
3. Once the phone was picked up, FXO port would off-hook to answer the PSTN call.